CHAPTER 4

Sound Waves in the Free Field

Practical acoustic problems are invariably associated with people, buildings, rooms, airplanes, automobiles, etc. These can generally be classified either as problems in physics (sound as a stimulus) or problems in psychophysics (sound as a perception), and often as both. Acoustical problems can be very complex in a physical sense, for example, thousands of reflected components might be involved or obscure temperature gradients might bend the sound in such a way as to affect the results. When acoustical problems involve human beings and their reactions, "complexity" takes on a whole new meaning.

Don't be discouraged if you want a practical understanding of acoustics, but your background is in another field, or you have little technical background at all. The inherent complexity of acoustics is pointed out only to justify going back to the inherent simplicity of sound in a free field as a starting point in the study of other types of practical sound fields.

Free Sound Field: Definition

Sound in a free field travels in straight lines, unimpeded and undeflected. Unimpeded sound is sound that is unreflected, unabsorbed,

undiffracted, unrefracted, undiffused, and not subjected to resonance effects. These are all hazards that could (and do) face a simple ray of sound leaving a source.

Free space must not be confused with cosmological space. Sound cannot travel in a vacuum; it requires a medium such as air. Here, free space means any air space in which sound acts as though it is in the theoretical free space. Limited free space can even exist in a room under very special conditions.

Sound Divergence

The point source of Fig. 4-1 radiates sound at a fixed power. This sound is of uniform intensity (power per unit area) in all directions. The circles represent spheres having radii in simple multiples. All of the sound power passing through the small square area at radius *d* also passes through the areas at *2d*, *3d*, *4d*, etc. This increment of the total sound power traveling in this single direction is spread over increasingly greater areas as the radius is increased. Intensity decreases with distance. As the area of a sphere is $4\pi r^2$, the area of a small segment on the surface of the sphere also varies as the square of the radius. Doubling the distance from *d* to *2d* reduces the intensity to $1/4$, tripling the distance reduces the intensity to $1/9$, and quadrupling the distance reduces intensity to $1/16$. *Intensity of sound is inversely proportional to the square of the distance in a free field.*

Intensity of sound (power per unit area) is a difficult parameter to measure. Sound pressure is easily measured. As intensity is proportional to the square of sound pressure, the *inverse square law* (for intensity) becomes *the inverse distance law* (for sound pressure). In other words, sound pressure varies inversely as the first power of the distance. In Fig. 4-2, the sound-pressure level in decibels is plotted against distance. This illustrates the basis for the common and very useful expression, *6 dB per doubling of the distance* that, again, applies only for a free field.

Examples: Free-Field Sound Divergence

When the sound-pressure level L_1 at distance d_1 from a point source is known, the sound-pressure level L_2 at another distance d_2 can be calculated from:

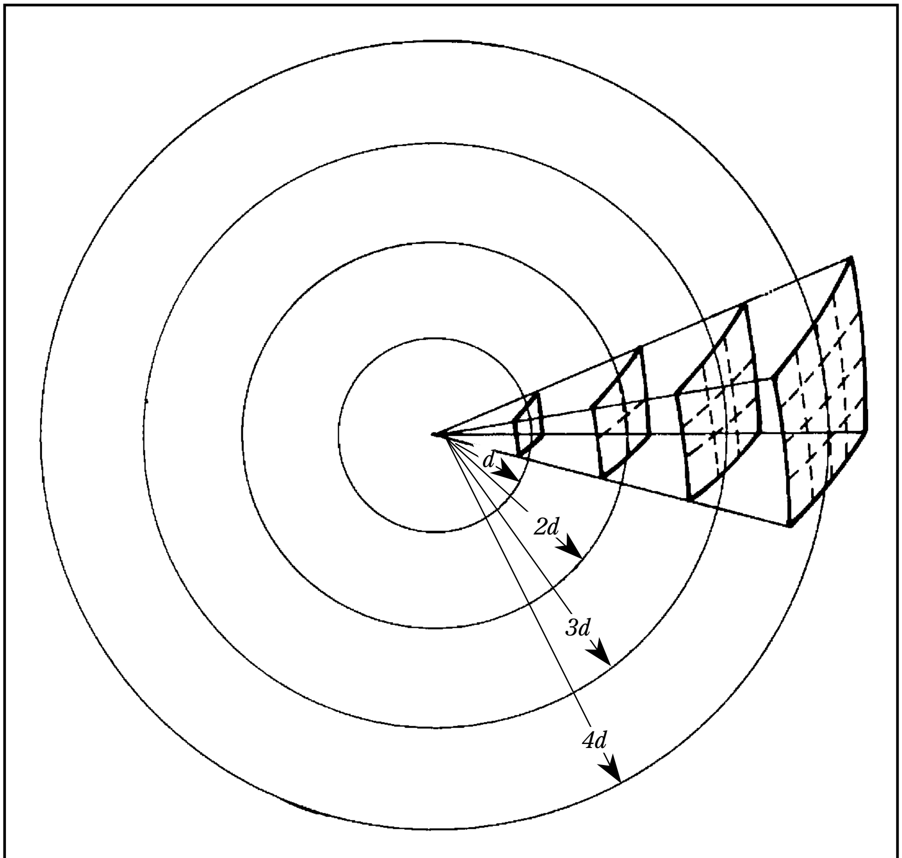

FIGURE 4-1

In the solid angle shown, the same sound energy is distributed over spherical surfaces of increasing area as *d* is increased. The intensity of sound is inversely proportional to the square of the distance from the point source.

$$L_2 = L_1 - 20 \log \frac{d_2}{d_1}, \text{ decibels} \qquad \textbf{(4-1)}$$

In other words, the difference in sound-pressure level between two points that are d_1 and d_2 distance from the source is:

$$L_2 - L_1 = 20 \log \frac{d_2}{d_1}, \text{ decibels} \qquad \textbf{(4-2)}$$

For example, if a sound-pressure level of 80 dB is measured at 10 ft, what is the level at 15 ft?

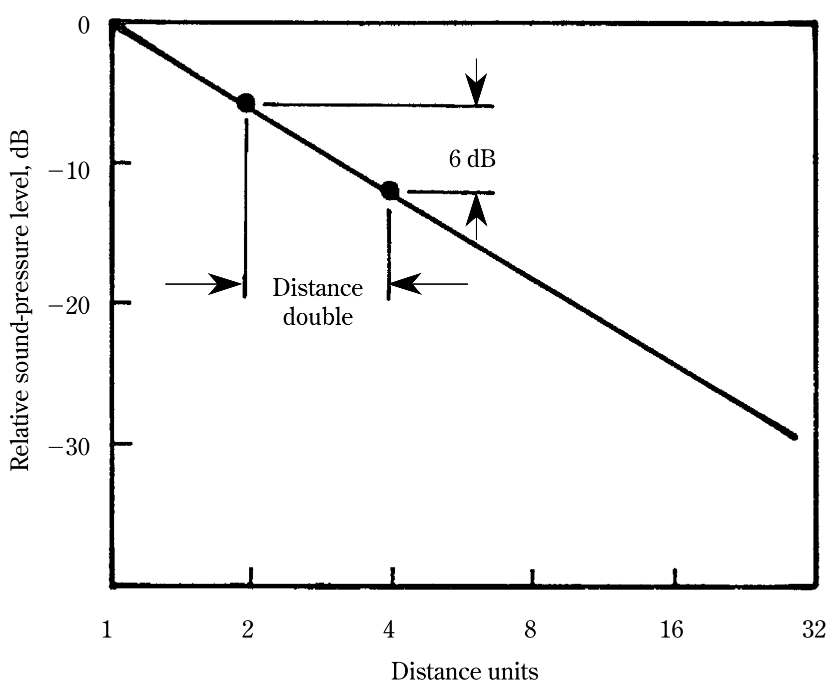

FIGURE 4-2

The inverse square law of sound intensity becomes the inverse distance law for sound pressure. This means that sound-pressure level is reduced 6 dB for each doubling of the distance.

Solution:

$$20 \log 10/15 = 3.5 \text{ dB; the level is } 80 - 3.5 = 76.5 \text{ dB.}$$

What is the sound-pressure level at 7 ft?

Solution:

$$20 \log 10/7 = +\ 3.1 \text{ dB, and level is } 80 + 3.1 = 83.1 \text{ dB.}$$

All this is for a free field in which sound diverges spherically, but this procedure may be helpful for rough estimates even under other conditions.

If a microphone is 5 feet from an enthusiastic soprano and the VU meter in the control room peaks +6, moving the microphone to 10 feet would bring the reading down *approximately* 6 dB. The word "approximately" is important. The inverse square law holds true only for free field conditions. The effect of sound energy reflected from

walls would be to make the change for a doubling of the distance something less than 6 dB.

An awareness of the inverse square law is of distinct help in estimating acoustical situations. For instance, a doubling of the distance from 10 to 20 feet would, for free space, be accompanied by the same sound-pressure level decrease, 6 dB, as for a doubling from 100 to 200 feet. This accounts for the great carrying power of sound outdoors.

Inverse Square in Enclosed Spaces

Free fields exist in enclosed spaces only in very special and limited circumstances. The reflections from the enclosing surfaces affect the way sound level decreases with distance. No longer does the inverse square law or the inverse distance law describe the entire sound field. For example, assume that there is an installed loudspeaker in an enclosed space that is capable of producing a sound-pressure level of 100 dB at a distance of 4 ft. As shown in the graph of Fig. 4-3, free field

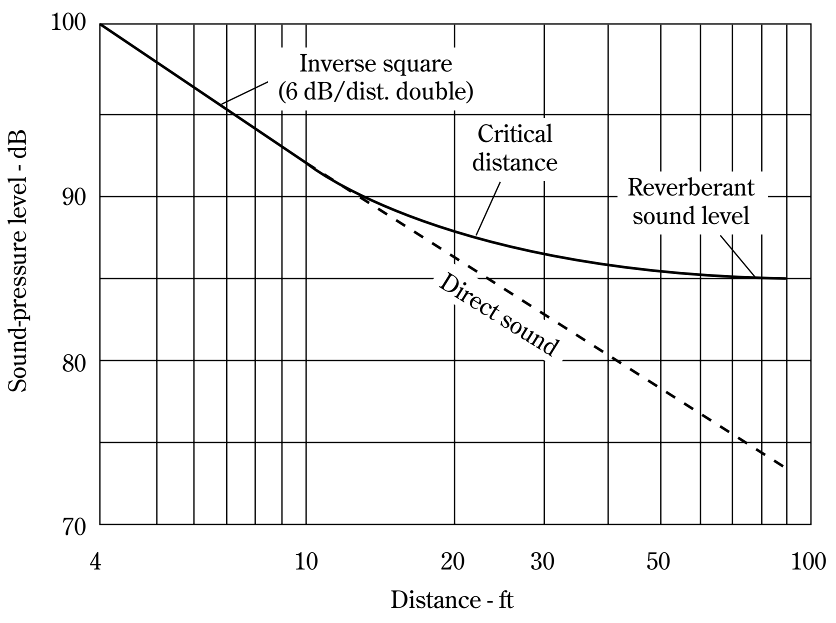

FIGURE 4-3

Even in an enclosed space the inverse square law is followed close to the source. By definition, the *critical distance* is that distance at which the direct sound pressure is equal to the reverberant sound pressure.

conditions exist close to the loudspeaker. This means that spherical divergence prevails in this limited space, and reflections from the surfaces are of negligible comparative level. Moving away from the loudspeaker, the effects of sound reflected from the surfaces of the room begin to be effective. At the *critical distance* the direct and the reflected sound are equal. The critical distance may be taken as a rough single-figure description of the acoustics of the environment.

In the region very close to the loudspeaker, the sound field is in considerable disarray. The loudspeaker, at such close distances, can in no way be considered a point source. This region is called the *near field*. Only after moving several loudspeaker dimensions away from it can significant measurements be made in the *far field*.

Hemispherical Propagation

True spherical divergence implies no reflecting surfaces at all. Tied to this earth's surface as we are, how about hemispherical sound propagation over the surface of this planet? Estimates made by the very convenient "6 dB per distance double" rule are only rough approximations.

Reflections from the surface of the earth outdoors usually tend to make the sound level with distance something less than that indicated by the 6 dB per distance double. The reflective efficiency of the earth's surface varies from place to place. Note the sound level of a sound at 10 ft and again at 20 ft from the source. The difference between the two will probably be closer to 4 dB than 6 dB. For such outdoor measurements the distance law must be taken at "X dB (4?, 5?) per distance double." There is also the effect of general environmental noise that can influence the measurement of specific sound sources.

CHAPTER 10

Reflection of Sound

If a sound is activated in a room, sound travels radially in all directions. As the sound waves encounter obstacles or surfaces, such as walls, their direction of travel is changed, i.e., they are reflected.

Reflections from Flat Surfaces

Figure 10-1 illustrates the reflection of waves from a sound source from a rigid, plane wall surface. The spherical wavefronts (solid lines) strike the wall and the reflected wavefronts (broken lines) are returned toward the source.

Like the light/mirror analogy, the reflected wavefronts act as though they originated from a *sound image*. This image source is located the same distance behind the wall as the real source is in front of the wall. This is the simple case—a single reflecting surface. In a rectangular room, there are six surfaces and the source has an image in all six sending energy back to the receiver. In addition to this, images of the images exist, and so on, resulting in a more complex situation. However, in computing the total sound intensity at a given receiving point, the contributions of all these images must be taken into consideration.

Sound is reflected from objects that are large compared to the wavelength of the impinging sound. This book would be a good

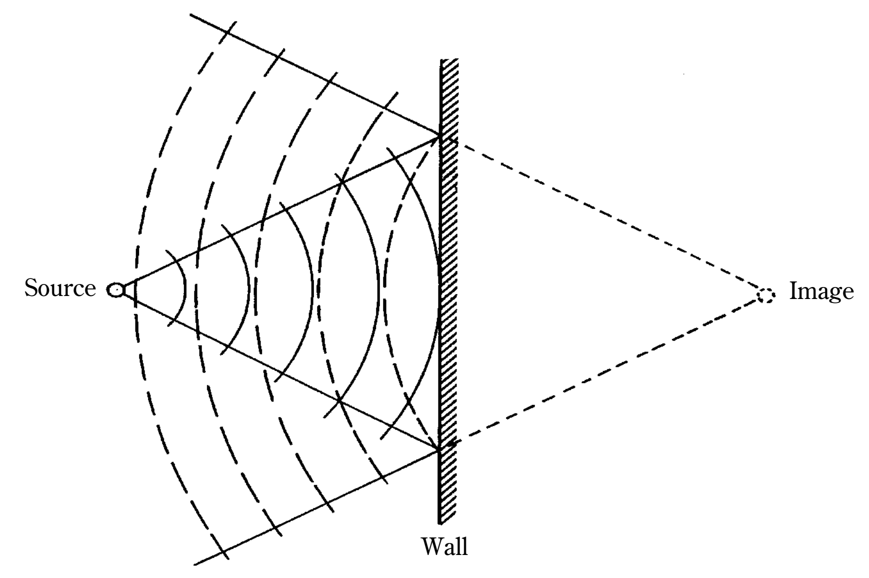

FIGURE 10-1

Reflection of sound from a point source from a flat surface (incident sound, solid lines; reflected sound, broken lines). The reflected sound appears to be from a virtual image source.

reflector for 10 kHz sound (wavelength about an inch). At the low end of the audible spectrum, 20 Hz sound (wavelength about 56 ft) would sweep past the book and the person holding it as though they did not exist, and without appreciable *shadows*.

Below 300–400 Hz, sound is best considered as waves (chapter 15 expounds on this). Sound above 300–400 Hz is best considered as traveling in rays. A ray of sound may undergo many reflections as it bounces around a room. The energy lost at each reflection results in the eventual demise of that ray. Even the ray concept is an oversimplification: Each ray should really be considered as a "pencil" of diverging sound with a spherical wavefront to which the inverse square law applies.

The mid/high audible frequencies have been called the *specular* frequencies because sound in this range acts like light rays on a mirror. Sound follows the same rule as light: The angle of incidence is equal to the angle of reflection, as in Fig. 10-2.

Doubling of Pressure at Reflection

The sound pressure on a surface normal to the incident waves is equal to the energy-density of the radiation in front of the surface. If the surface is a perfect absorber, the pressure equals the energy-density of the incident radiation. If the surface is a perfect reflector, the pressure equals the energy-density of both the incident and the reflected radiation. Thus the pressure at the face of a perfectly reflecting surface is twice that of a perfectly absorbing surface. At this point, this is only an interesting sidelight. In the study of standing waves in Chap. 15, however, this pressure doubling takes on greater significance.

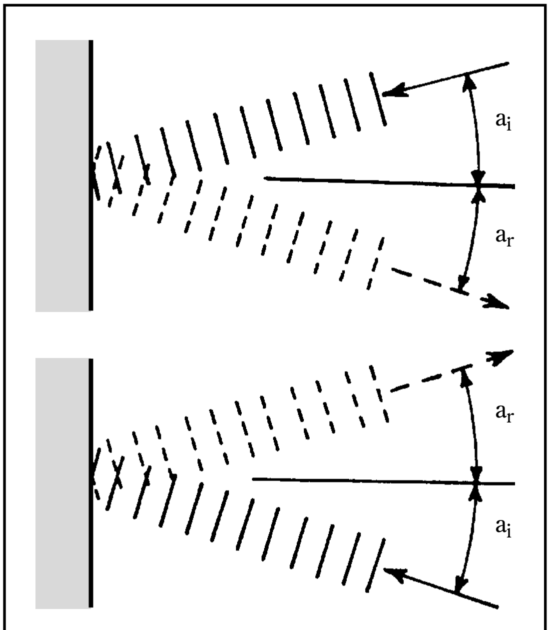

FIGURE 10-2

The angle of incidence, a_i, is equal to the angle of reflection, a_r.

Reflections from Convex Surfaces

Spherical wavefronts from a point source tend to become plane waves at greater distance from the source. For this reason impinging sound on the various surfaces to be considered will be thought of as plane wavefronts. Reflection of plane wavefronts of sound from a solid convex surface tends to scatter the sound energy in many directions as shown in Fig. 10-3. This amounts to a diffusion of the impinging sound.

The polycylindrical sound-absorbing system described in the previous chapter both absorbs sound and contributes to much-needed diffusion in the room by reflection from the cylindrically shaped surface.

Reflections from Concave Surfaces

Plane wavefronts of sound striking a concave surface tend to be focussed to a point as illustrated on Fig. 10-4. The precision with which sound is focussed to a point is determined by the shape of the

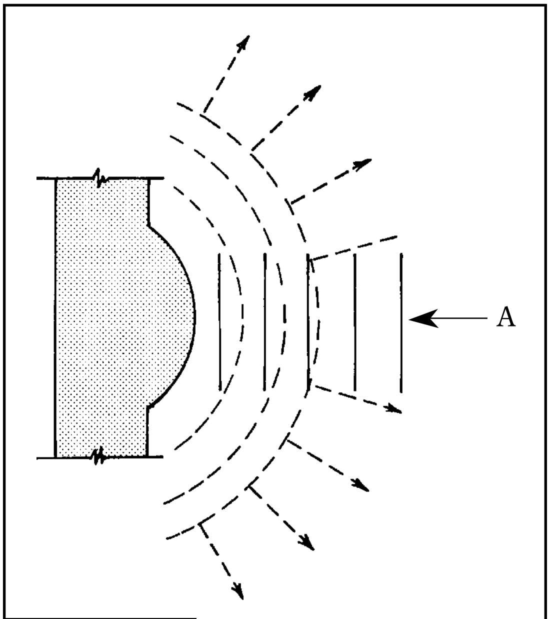

FIGURE 10-3

Plane sound waves impinging on a convex irregularity tend to be dispersed through a wide angle if the size of the irregularity is large compared to the wavelength of the sound.

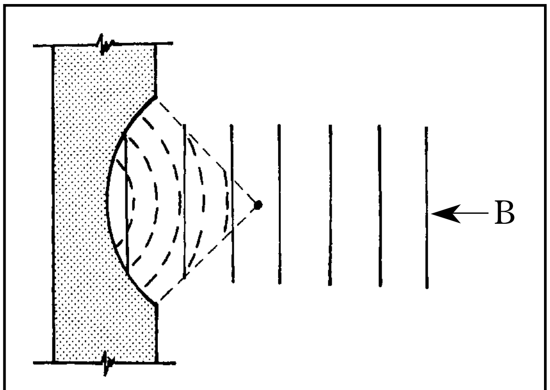

FIGURE 10-4

Plane sound waves impinging on a concave irregularity tend to be focussed if the size of the irregularity is large compared to the wavelength of the sound.

concave surface. Spherical concave surfaces are common because they are readily formed. They are often used to make a microphone highly directional by placing it at the focal point. Such microphones are frequently used to pick up field sounds at sporting events or in recording songbirds or other animal sounds in nature. In the early days of broadcasting sporting events in Hong Kong, a resourceful technician saved the day by using an ordinary Chinese wok, or cooking pan, as a reflector. Aiming the microphone into the reflector at the focal point provided an emergency directional pickup. Concave surfaces in churches or auditoriums can be the source of serious problems as they produce concentrations of sound in direct opposition to the goal of uniform distribution of sound.

The effectiveness of reflectors for microphones depends on the size of the reflector with respect to the wavelength of sound. A 3-ft-diameter spherical reflector will give good directivity at 1 kHz (wavelength about 1 ft), but it is practically nondirectional at 200 Hz (wavelength about 5.5 ft).

Reflections from Parabolic Surfaces

A parabola has the characteristic of focusing sound precisely to a point (Fig. 10-5). It is generated by the simple equation $y = x^2$. A very "deep" parabolic surface, such as that of Fig. 10-5, exhibits far better directional properties than a shallow one. Again, the directional properties

depend on the size of the opening in terms of wavelengths. Figure 10-5 shows the parabola used as a directional sound source with a small, ultrasonic Galton Whistle pointed inward at the focal point.

Plane waves striking such a reflector would be brought to a focus at the focal point. Conversely, sound emitted at the focal point of the parabolic reflector generates plane wavefronts. This is demonstrated in the photographs of Figs. 10-6 and 10-7 in which standing waves are produced by reflections from a heavy glass plate. The force exerted by the vibration of the air particles on either side of a node is sufficient to hold slivers of cork in levitation.

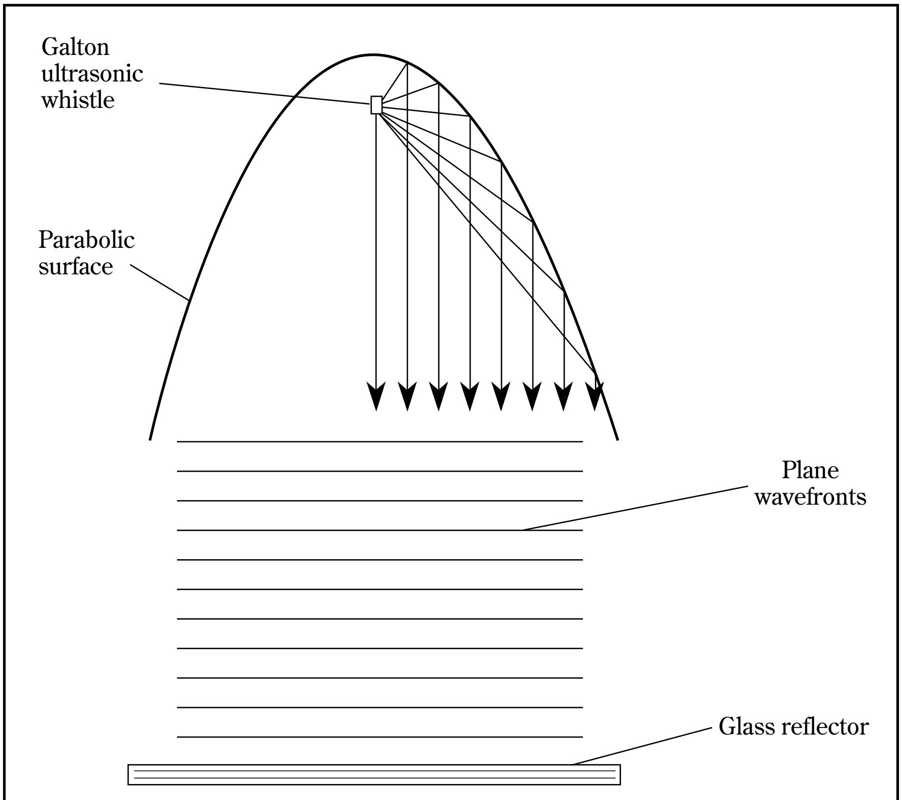

FIGURE 10-5

A parabolic surface can focus sound precisely at a focal point or, the converse, a sound source placed at the focal point can produce plane, parallel wavefronts. In this case, the source is an ultrasonic Galton Whistle blown by compressed air with the results shown in Figs. 10-6 and 10-7.

Reflections Inside a Cylinder

St. Paul's Cathedral in London boasts a *whispering gallery*. The way this whispering gallery works is explained in the diagram of Fig. 10-8. Reflections from the exterior surfaces of cylindrical shapes have been mentioned in the treatment of "polys." In this case the source and receiver are both inside a mammoth, hard-surfaced cylindrical room.

At the source, a whisper directed tangentially to the surface is clearly heard on the receiver side. The phenomenon is assisted by the fact that the walls are dome-shaped. This means that upward-directed components of the whispered sounds tend to be reflected downward and conserved rather than lost above.

Standing Waves

The concept of standing waves is directly dependent on the reflection of sound as emphasized in Chap. 15. Assume two flat, solid parallel walls separated a given distance. A sound source between them radiates sound of a specific frequency. The wavefront striking the right wall is reflected back toward the source, striking the left wall where it is again reflected back toward the right wall, and so on. One wave travels to the right, the other toward the left. The two traveling waves interact to form a standing wave. Only the standing wave, the interaction of the two, is stationary. The frequency of the radiated sound is such as to establish this resonant condition between the wavelength of the sound and the distance between the two surfaces. The pertinent point at the moment is that this phenomenon is entirely dependent on the reflection of sound at the two parallel surfaces.

Reflection of Sound from Impedance Irregularities

The television repairman is concerned about matching the electrical impedance of the television receiver to that of the transmission line, and matching the transmission line to the impedance of the antenna (or cable). Mismatches of impedance give rise to reflections, which cause numerous undesirable effects.

FIGURE 10-6

A parabolic reflector following the equation $y=x^2$ reflects sound from an ultrasonic Galton Whistle to form a stable standing wave system capable of levitating bits of cork.

FIGURE 10-7

Close-up of the levitated cork chips of Fig. 10-6. If the cork chips are spaced about ½ inch, the frequency of the sounds emitted by the Galton Whistle is about 30 kHz.

A similar situation prevails in an air-conditioning duct. A sound wave (noise) traveling in the duct suddenly encounters the large open space of the room. This discontinuity (impedance mismatch) reflects a significant portion of the sound (fan noise, etc.) back toward the

source. This is an example of a benevolent mismatch as the air-conditioner noise is reduced in the room.

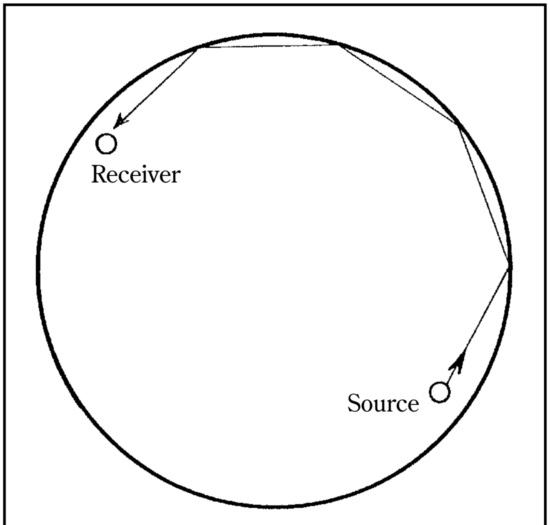

FIGURE 10-8

Graphic explanation of the "Whispering Gallery" of St. Paul's Cathedral, London. A whisper directed tangentially to the cylinderical surface is readily heard by the receiver on the far side of the room.

The Corner Reflector

In an art museum with large Dutch paintings on display, the eyes of certain subjects seem to follow as one walks by. Corner reflectors are like that. There seems to be no way of escaping their pernicious effect. The corner reflector of Fig. 10-9, receiving sound from the source *S*, sends a reflection directly back toward the source. If the angles of incidence and reflection are carefully noted, a source at *B* will also send a direct, double-surface reflection returning to the source. A source at *C*, on the opposite of *B*, is subject to the same effect.

You might be instinctively aware of perpendicular (normal) reflections from surrounding walls, but now consider reflections from the four corners of the room that follow the source around the room. Corner reflections suffer losses at two surfaces, tending to make them somewhat less intense than normal reflections at the same distance.

The corner reflector of Fig. 10-9 involves only two surfaces. How about the four upper tri-corners of the room formed by ceiling and walls and another four formed by floor and wall surfaces? The same follow-the-source principle applies. In fact, sonar and radar people have long employed targets made of three circular plates of reflecting material assembled so that each is perpendicular to the others.

Echo-Sounding

Objects can be located by sending out a pulse of sound and noting the time it takes for the reflected echo to return. Directional sources of sound make possible the determination of both the azimuth angle and the distance to the reflecting object. This principle has been widely applied in water depth sounders, sonar on submarines, etc. All

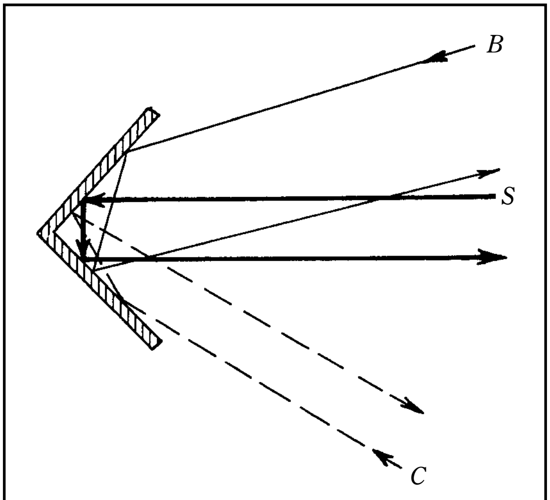

FIGURE 10-9

The corner reflector has the property of reflecting sound back toward the source from any direction.

depend on the reflection of sound from the bottom of the ocean, or enemy target.

Perceptive Effects of Reflections

In the reproduction of sound in a high-fidelity listening room or control room of a recording studio, the sound of the loudspeakers falling on the ear of the listener is very much affected by reflections from the surfaces of the room. This is another manifestation of sound reflection. A comprehensive consideration of human perception of such reflections is included in Chap. 16.

CHAPTER 11

Diffraction of Sound

It is well known that sound travels around corners and around obstacles. Music reproduced in one room of a home can be heard down the hall and in other rooms. Diffraction is one of the mechanisms involved in this. The character of the music heard in distant parts of the house is different. In distant rooms the bass notes are more prominent because their longer wavelengths are readily diffracted around corners and obstacles.

Rectilinear Propagation

Wavefronts of sound travel in straight lines. Sound rays, a concept applicable at mid/high audible frequencies, are considered to be pencils of sound that travel in straight lines perpendicular to the wavefront. Sound wavefronts and sound rays travel in straight lines, except when something gets in the way. Obstacles can cause sound to be changed in its direction from its original rectilinear path. The process by which this change of direction takes place is called *diffraction*.

Alexander Wood, the early Cambridge acoustician, recalled Newton's pondering over the relative merits of the corpuscular and wave theories of light. Newton finally decided that the corpuscular theory was the correct one because light is propagated rectilinearly. Later it was demonstrated that light is not always propagated rectilinearly,

that diffraction can cause light to change its direction of travel. In fact, all types of wave motion, including sound, are subject to diffraction.

The shorter the wavelength (the higher the frequency), the less dominant is the phenomenon of diffraction. Diffraction is less noticeable for light than it is for sound because of the extremely short wavelengths of light. Obstacles capable of diffracting (bending) sound must be large compared to the wavelength of the sound involved. The well-worn example of ocean waves is still one of the best. Ocean waves sweep past a piling of a dock with scarcely a disturbance. Ocean waves, however, are bent around an end of an island.

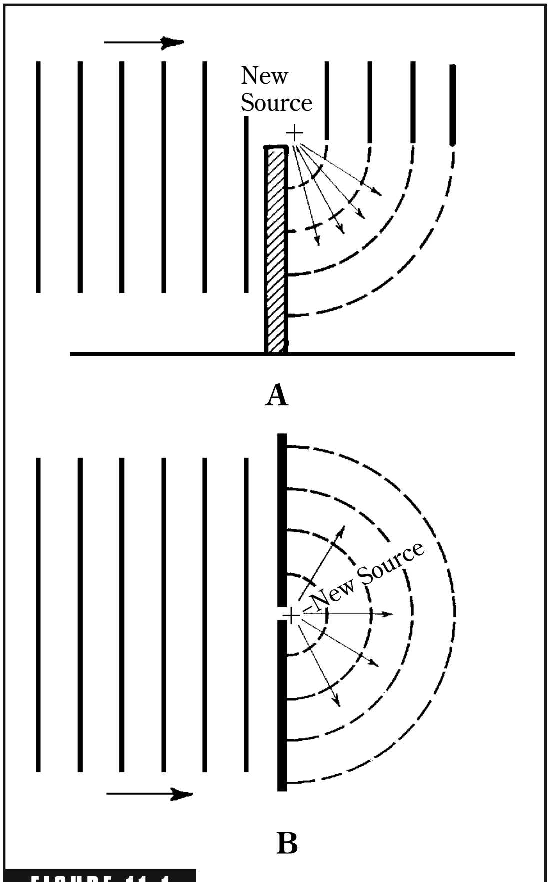

FIGURE 11-1

(A) If the brick wall is large in terms of the wavelength of the sound, the edge acts as a new source, radiating sound into the shadow zone. (B) Plane waves of sound impinging on the heavy plate with a small hole in it sets up spherical wavefronts on the other side due to diffraction of sound.

Diffraction and Wavelength

The effectiveness of an obstacle in diffracting sound is determined by the acoustical size of the obstacle. Acoustical size is measured in terms of the wavelength of the sound. One way of looking at the illustration shown later, Fig. 11-3, is that the obstacle in B is the same physical size as that of A, but the frequency of the sound of A is one tenth that of B. If the obstacle in B is 1 ft long and that of A 0.1 ft long, the frequency of the sound in A could well be 1,000 Hz (wavelength 1.13 ft), and that of B could be 100 Hz (wavelength 11.3 ft). The same drawing could be used if the obstacle of A were 0.01 ft long with a frequency of 10,000 Hz (wavelength 0.113 ft) and the obstacle of B were 0.1 ft long with a frequency of 1,000 Hz (wavelength 1.13 ft).

In Fig. 11-1, two types of obstructions to plane wavefronts of sound are depicted. In Fig. 11-1A a heavy brick wall is the obstacle. The sound waves are reflected from the

face of the wall, as expected. The upper edge of the wall acts as a new, virtual source sending sound energy into the "shadow" zone behind the wall by diffraction. The mechanism of this effect will be considered in more detail later in this chapter.

In Fig. 11-1B the plane wavefronts of sound strike a solid barrier with a small hole in it. Most of the sound energy is reflected from the wall surface, but that tiny portion going through the hole acts as a virtual point source, radiating a hemisphere of sound into the "shadow" zone on the other side.

Diffraction of Sound by Large and Small Apertures

Figure 11-2A illustrates the diffraction of sound by an aperture that is many wavelengths wide. The wavefronts of sound strike the heavy obstacle: some of it is reflected, some goes right on through the wide aperture. The arrows indicate that some of the energy in the main beam is diverted into the shadow zone. By what mechanism is this diversion accomplished?

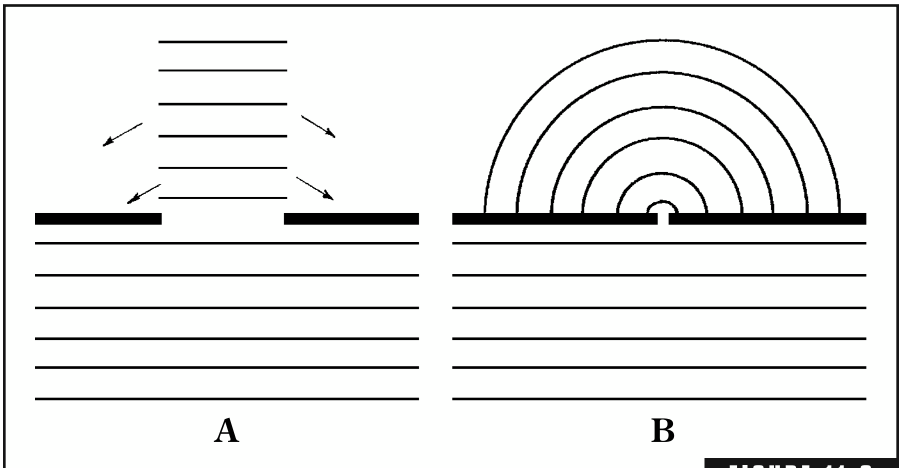

FIGURE 11-2

(A) An aperature large in terms of wavelength of sound allows wavefronts to go through with little disturbance. These wavefronts act as lines of new sources radiating sound energy into the shadow zone. (B) If the aperature is small compared to the wavelength of the sound, the small wavefronts which do penetrate the hole act almost as point sources, radiating a hemispherical field of sound into the shadow zone.

For an answer, the work of Huygens is consulted.[1] He enunciated a principle that is the basis of very difficult mathematical analyses of diffraction. The same principle also gives a simple explanation of how sound energy is diverted from the main beam into the shadow zone. Huygens' principle can be paraphrased as:

Every point on the wavefronts of sound that has passed through an aperture or passed a diffracting edge is considered a point source radiating energy back into the shadow zone.

The sound energy at any point in the shadow zone can mathematically be obtained by summing the contributions of all of these point sources on the wavefronts.

In Fig. 11-2A, each wavefront passing through the aperture becomes a row of point sources radiating diffracted sound into the shadow zone. The same principle holds for Fig. 11-2B except that the aperture is very small and only a small amount of energy passes through it. The points on the limited wavefront going through the hole are so close together that their radiations take the form of a hemisphere.

Diffraction of Sound by Obstacles

In Fig. 11-3A the obstacle is so small compared to the wavelength of the sound that it has no appreciable effect on the passage of sound. In Fig. 11-3B, however, the obstacle is many wavelengths long and it has a definite effect in casting a shadow behind the obstacle. Each wavefront passing the obstacle becomes a line of new point sources radiating sound into the shadow zone.

A very common example of an obstacle large compared to the wavelength of the impinging sound is the highway noise barrier shown in Fig. 11-4. If the wavelength of the impinging sound is indicated by the spacing of the spherical wavefronts hitting the barrier, the barrier size is acoustically great. At higher frequencies the barrier becomes even *larger*, and at lower frequencies it becomes acoustically *smaller*. First, the sound reflected from the wall must be noted. It is as though the sound were radiated from a virtual image on the far side of the wall. The wavefronts passing the top edge of the wall can be considered as lines of point sources radiating sound. This is the source of the sound penetrating the shadow zone.

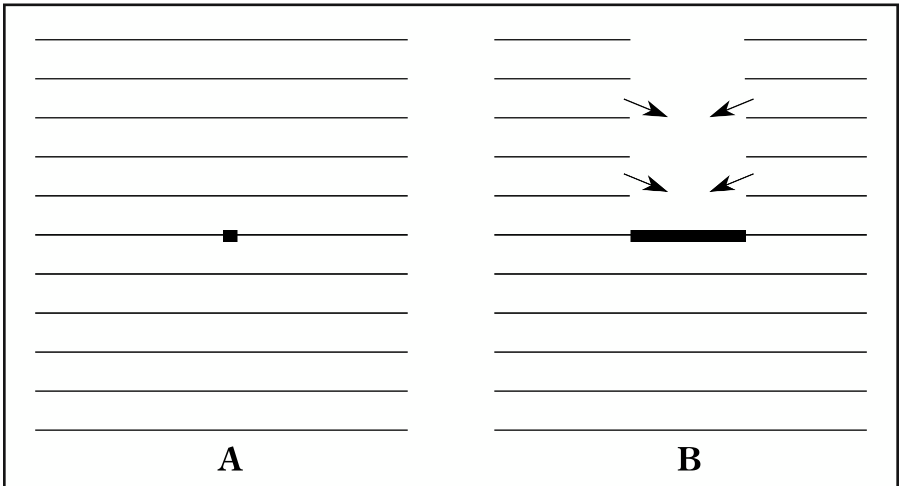

FIGURE 11-3

(A) An obstacle very much smaller than the wavelength of sound allows the wavefronts to pass essentially undisturbed. (B) An obstacle large compared to the wavelength of sound casts a shadow that tends to be irradiated from sources on the wavefronts of sound that go past the obstacle.

Figure 11-5 gives some idea of the effectiveness of highway barriers and of the intensity of the sound in the shadow of a high, massive wall. The center of the highway is taken to be 30 ft from the wall on one side, and the home or other sensitive area is considered to be 30 ft on the other side of the wall (the shadow side). A wall 20 ft high yields something like 25 dB of protection from the highway noise at 1,000 Hz. At 100 Hz, the attenuation of the highway noise is only about 15 dB. At the higher audible frequencies, the wall is more effective than at lower frequencies. The shadow zone behind the wall tends to be shielded from the high-frequency components of the highway noise. The low-frequency components penetrate the shadow zone by diffraction.

Diffraction of Sound by a Slit

Figure 11-6 diagrams a classical experiment performed by Pohl in acoustical antiquity and described by Wood[1] in somewhat more recent antiquity. One must admire the precise results obtained with crude measuring instruments (high-pitched whistle, sound radiometer). The

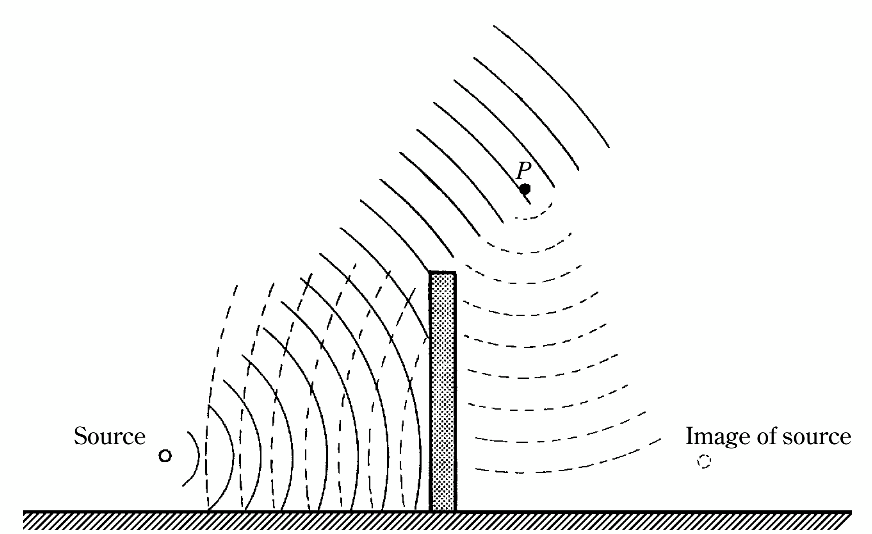

FIGURE 11-4

The classic sound barrier case. The sound striking the wall is reflected as though the sound is radiated from a virtual image of the source. That sound passing the top edge of the wall acts as though the wavefronts are lines of sources radiating sound energy into the shadow zone.

equipment layout of Fig. 11-6A is very approximate. Actually the source/slit arrangement rotated about the center of the slit and the measuring radiometer was at a distance of 8 meters. The slit width was 11.5 cm wide, the wavelength of the measuring sound was 1.45 cm (23.7 kHz). The graph of Fig. 11-6B shows the intensity of the sound versus the angle of deviation. The dimension B indicates the geometrical boundaries of the ray. Anything wider that B is caused by diffraction of the beam by the slit. A narrower slit would yield correspondingly more diffraction and a greater width of the beam. The increase in width of the beam is the striking feature of this experiment.

Diffraction by the Zone Plate

The zone plate can be considered an acoustic lens. It consists of a circular plate with a set of concentric, annular slits of cunningly devised radii. If the focal point is at a distance of r from the plate,

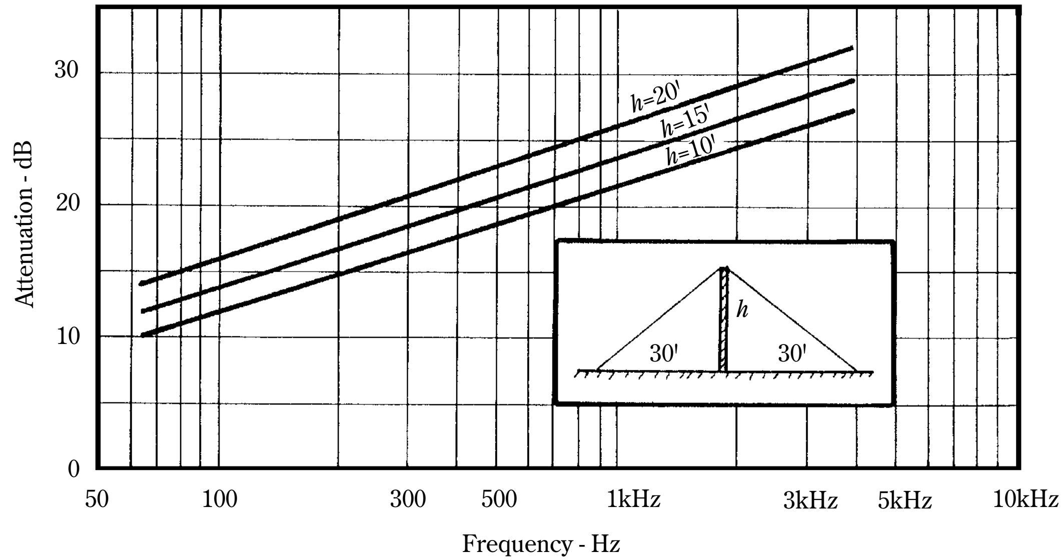

FIGURE 11-5

An estimation of the effectiveness of a sound barrier in terms of sound (or noise) attentuation as a function of frequency and barrier height. *(After Rettinger.[4])*

the next longer path must be $r + \lambda/2$ where λ is the wavelength of the sound falling on the plate from the source. Successive path lengths are $r + \lambda$, $r + 3/2\lambda$, and $r + 2\lambda$. These path lengths differ by $\lambda/2$, which means that the sound through all the slits will arrive at the focal point in phase which, in turn, means that they add constructively, intensifying the sound.[2] See Fig. 11-7.

Diffraction around the Human Head

Figure 11-8 illustrates the diffraction caused by a sphere roughly the size of the human head. This diffraction by the head as well as reflections and diffractions from the shoulders and the upper torso influences human perception of sound. In general, for sound of frequency 1–6 kHz arriving from the front, head diffraction tends to increase the sound pressure in front and decrease it behind the head. For frequencies in the lower range the directional pattern tends to become circular.[2,3]

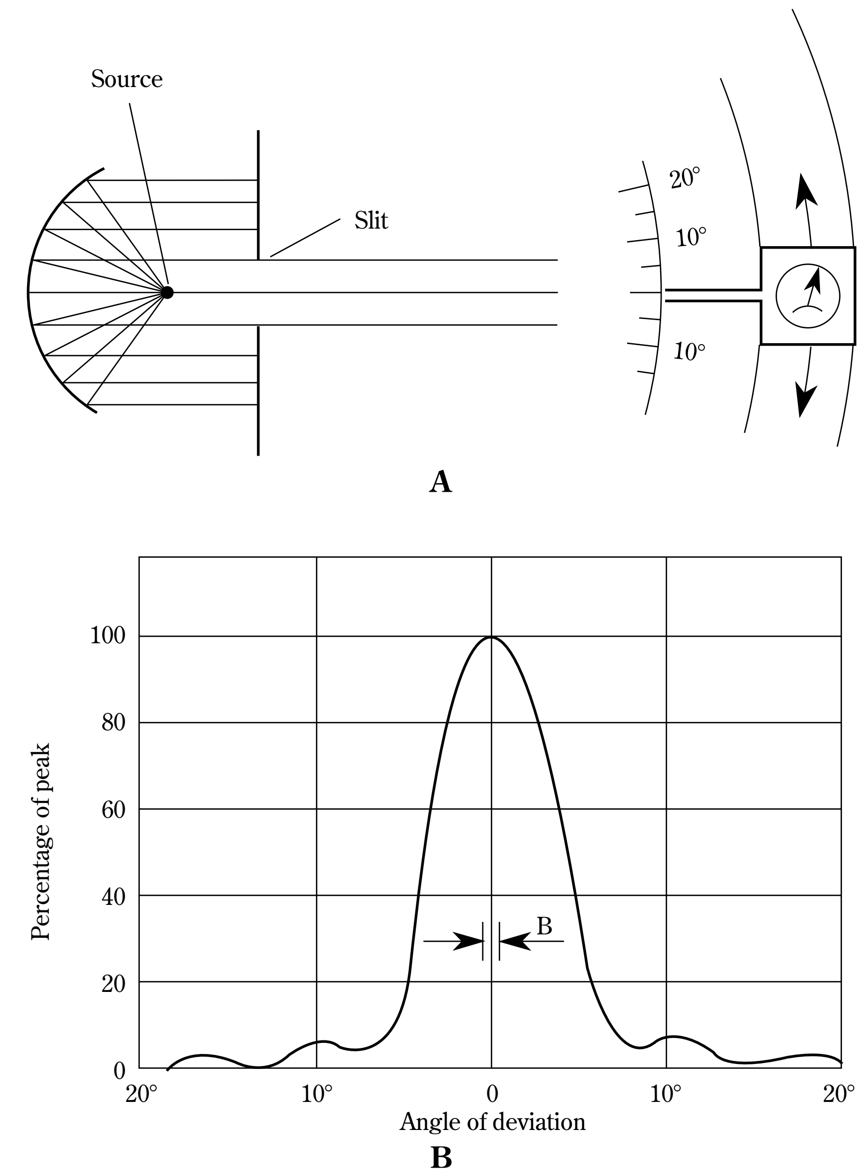

FIGURE 11-6

A consideration of Pohl's classic experiment in diffraction. (A) A very approximate suggestion of the equipment arrangement (see text). (B) The broadening of the beam B by diffraction. The narrower the slit the greater this broadening of the beam. *(After Wood.[1])*

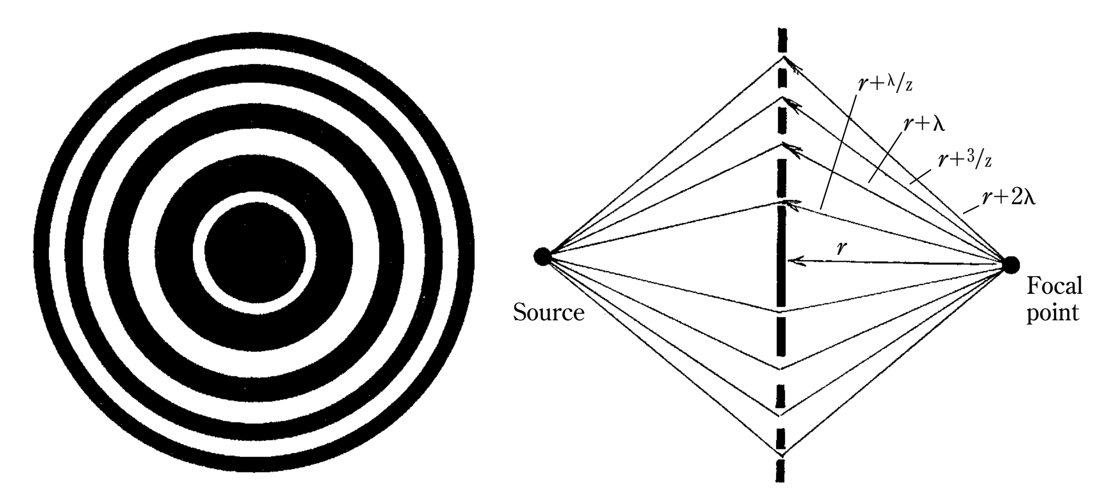

FIGURE 11-7

The zone plate or acoustic lens. The slits are so arranged that the several path lengths differ by multiples of a half wavelength of the sound so that all diffracted rays arrive at the focal point in phase, combining constructively. *(After Olson.[2])*

Diffraction by Loudspeaker Cabinet Edges

Loudspeaker cabinets are notorious for diffraction effects. If a loudspeaker is mounted near a wall and aimed away from the wall, the wall is still illuminated with sound diffracted from the corners of the box. Reflections of this sound can affect the quality of the sound at the listener's position. Measurements of this effect have been scarce, but Vanderkooy[5] and Kessel[6] have recently computed the magnitude of loudspeaker cabinet edge diffraction. The computations were made on a box loudspeaker with front baffle having the dimensions 15.7 × 25.2 in and depth of 12.6 in (Fig. 11-9). A point source of sound was located symmetrically at the top of the baffle. The sound from this point source was computed at a distance from the box. The sound arriving at the observation point is the combination of the direct sound plus the edge diffraction. This combination is shown in Fig. 11-10. Fluctuations due to edge diffraction for this particular typical situation approached plus or minus 5 dB. This is a significant change in overall frequency response of a reproduction system.

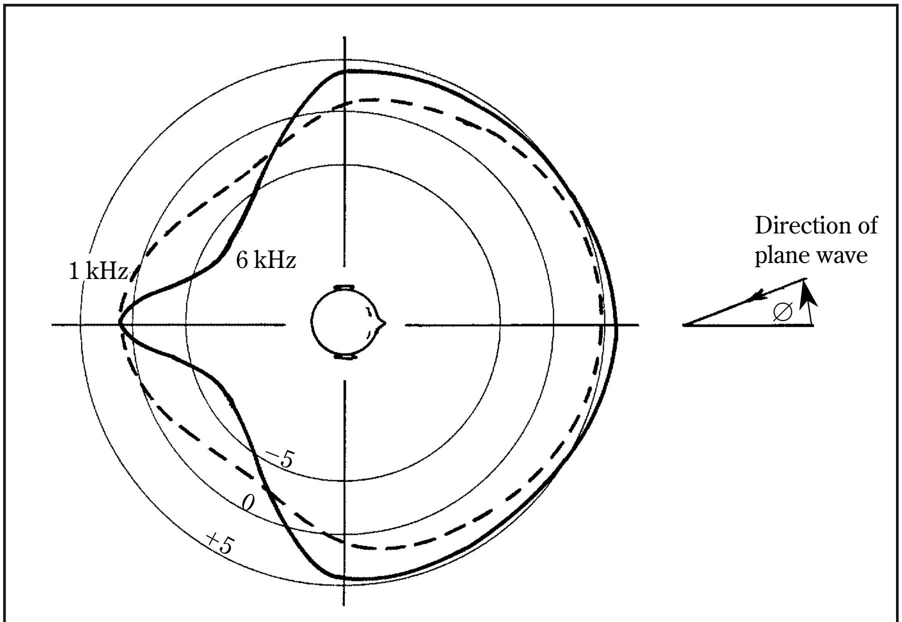

FIGURE 11-8

Diffraction around a solid sphere about the size of a human head. For sound in the 1-6 kHz range, sound pressure is generally increased in the front hemisphere and generally reduced in the rear. *(After Muller, Black and Davis, as reported by Olson.[2])*

This effect can be controlled (eliminated?) by setting the loudspeaker box face flush in a much larger baffling surface. There is also the possibility of rounding edges and the use of foam or "fuzz".[7]

Diffraction by Various Objects

Sound level meters were, in early days, boxes with a microphone protruding. Diffraction from the edges and corners of the box seriously affected the calibration of the microphones. Modern sound level meters have carefully rounded contours with the microphone mounted on a smooth, slender, rounded neck.

Diffraction from the casing of a microphone can cause deviations from the desired flat sensitivity.

In the measurement of sound absorption in large reverberation chambers, the common practice is to place the material to be measured

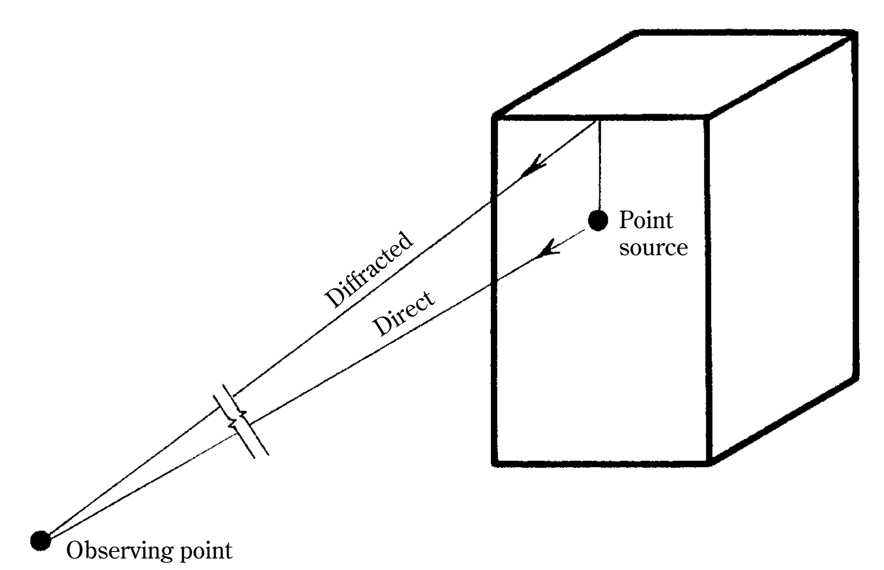

FIGURE 11-9

Arrangement for Vanderkooy's calculation of loudspeaker cabinet edge diffraction, shown in Fig. 11-10.

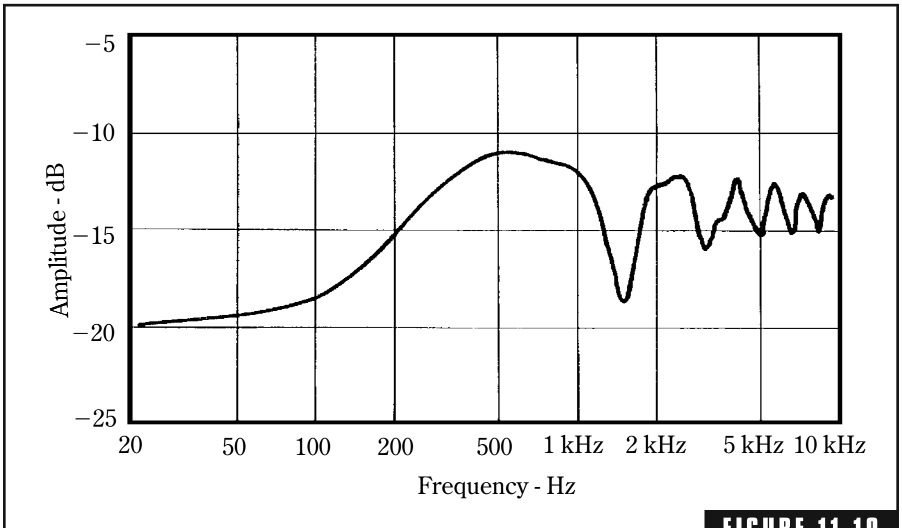

FIGURE 11-10

The calculated effects of loudspeaker edge diffraction on the direct signal in the arrangement of Fig.11-9. *(After Vanderkooy,[5] and Kessel.[6])*

in a 8 × 9-ft frame on the floor. Diffraction from the edges of this frame often result in absorption coefficients greater than unity. In other words, diffraction of sound makes the sample appear larger than it really is.

Small cracks around observation windows or back-to-back microphone or electrical service boxes in partitions can destroy the hoped-for isolation between studios or between studio and control room. The sound emerging on the other side of the hole or slit is spread in all directions by diffraction.

In summary, diffraction causes sound, which normally travels rectilinearly, to travel in other directions.

Endnotes

[1]Wood, Alexander, *Acoustics*, New York, Interscience Publishers, Inc. (1941).

[2]Olson, Harry F., *Elements of Acoustical Engineering*, New York, D. Van Nostrand Co. (1940).

[3]Muller, C.G., R. Black, and T.E. Davis, *The Diffraction Produced by Cylinders and Cubical Obstacles and by Circular and Square Plates*, J. Acous. Soc. Am., 10, 1 (1938) p. 6.

[4]Rettinger, M., *Acoustic Design and Noise Control*, Chemical Publishing Co. (1973).

[5]Vanderkooy, John, *A Simple Theory of Cabinet Edge Diffraction*, J. Audio Eng. Soc., 39, 12 (1991) 923-933.

[6]Kessel, R.T., *Predicting Far-Field Pressures from Near-Field Loudspeaker Measurements*, J. Audio Eng. Soc., Abstract, Vol. 36, p.1,026 (Dec 1988), preprint 2729.

[7]Kaufman, Richard J., *With a Little Help from My Friends*, AUDIO, 76, 9 (Sept 1992) 42-46.

CHAPTER 12

Refraction of Sound

About the turn of this century Lord Rayleigh was puzzled because some very powerful sound sources, such as cannon fire, could be heard only short distances some times and very great distances at other times. He set up a powerful siren that required 600 hp to maintain it. He calculated that if all this power were converted into energy as sound waves and spread uniformly over a hemisphere, how far could it be heard? Knowing the minimum audible intensity (10^{-16} watts per sq cm), his calculations indicated that the sound should be audible to a distance of 166,000 miles, more than 6 times the circumference of the earth!

It is indeed fortunate that such sound propagation is never experienced and that a range of a few miles is considered tops. There are numerous reasons why sound is not heard over greater distances. For one thing, the efficiency of sound radiators is usually quite low; not much of that 600 hp was actually radiated as sound. Energy is also lost as wavefronts drag across the rough surface of the earth. Another loss is dissipation in the atmosphere, but this is known to be very small. The result of such calculations and early experiments that fell far short of expectations served only to accelerate research on the effects of temperature and wind gradients on the transmission of sound.

Refraction of Sound

Refraction changes the direction of travel of the sound by differences in the velocity of propagation. *Diffraction* is changing the direction of travel of sound by encountering sharp edges and physical obstructions (chapter 11). Most people find it easy to distinguish between *absorption* and *reflection* of sound, but there is often confusion between *diffraction* and *refraction* (and possibly *diffusion*, the subject of the next chapter). The similarity of the sound of the words might be one cause for this confusion, but the major reason is the perceived greater difficulty of understanding diffraction, refraction, and diffusion compared to absorption and reflection. Hopefully Chaps. 9, 10, 11, 13, and this chapter will help to equalize and advance understanding of these five important effects.

Figure 12-1 recalls a very common observation of the apparent bending of a stick as one end touches the water surface or is actually immersed. This is an illustration of refraction of light. As the present subject is refraction of sound, which is another wave phenomenon, the relative refractive indices of air and water will be passed over.

Refraction of Sound in Solids

Figure 12-2 illustrates sound passing from a dense solid medium to a less dense medium. The sound speed in the denser medium is greater

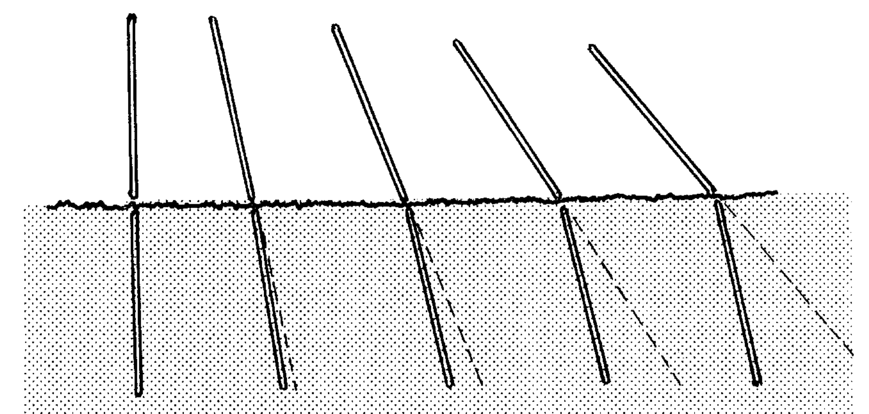

FIGURE 12-1

Touching a stick to the water surface illustrates refraction of light. Sound is another wave phenomenon that is also refracted by changes in media sound speed.

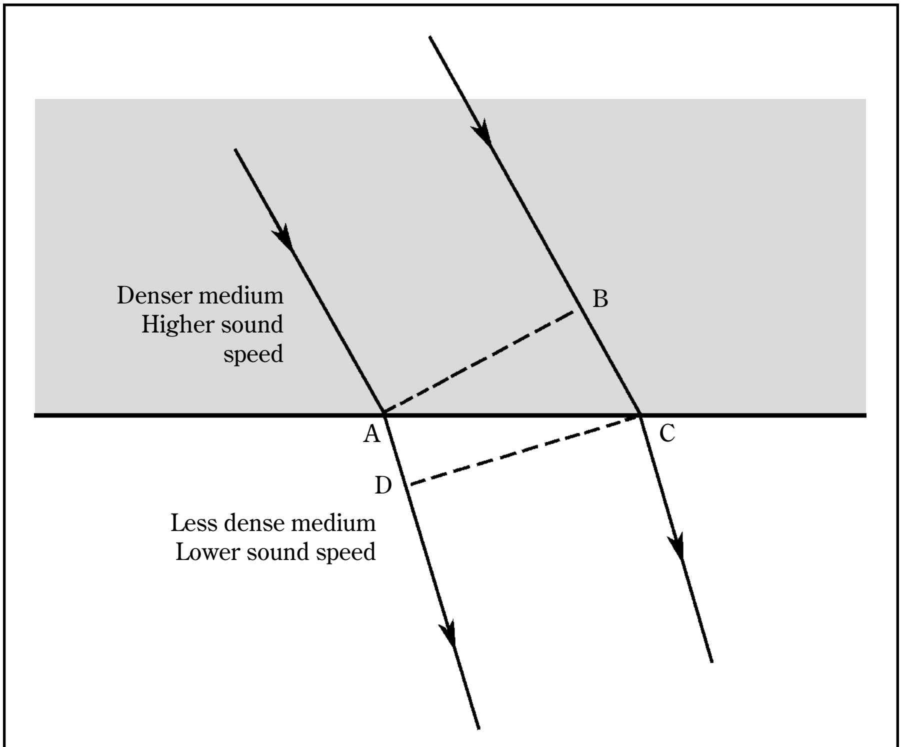

FIGURE 12-2

Rays of sound traveling from a denser medium having a certain sound speed into a less dense medium having a lower sound speed. The wavefront *AB* is not parallel to wavefront *DC* because the direction of the wave is changed due to refraction.

than that in the less dense one (Table 12-1). As one ray reaches the boundary between the two media at A, the other still has some distance to go. In the time it takes one ray to travel from B to C, the other ray has traveled a shorter distance from A to D in the new medium. Wavefront A-B represents one instant of time as does wavefront D-C an instant later. But these two wavefronts are no longer parallel. The rays of sound have been refracted at the interface of the two media having unlike sound speeds.

An analogy may assist memory and logic. Assume that the shaded area is paved and that the lower-density area is ploughed. Assume also that the wavefront A-B is a line of soldiers. The line of soldiers A-B, marching in military order, has been making good progress on the

Table 12-1. Speed of sound.

Medium	Speed of sound Ft/sec	Meters/sec
Air	1,130	344
Sea water	4,900	1,500
Wood, fir	12,500	3,800
Steel bar	16,600	5,050
Gypsum board	22,300	6,800

pavement. As soldier *A* reaches the ploughed ground he or she slows down and begins plodding over the rough surface. Soldier *A* travels to *D* on the ploughed surface in the same time that soldier *B* travels the distance *BC* on the pavement. This tilts the wavefront off in a new direction, which is the definition of refraction. In any homogeneous medium, sound travels rectilinearly (in the same direction). If a medium of another density is encountered, the sound is refracted.

Refraction of Sound in the Atmosphere

The atmosphere is anything but a stable, uniform medium for the propagation of sound. Sometimes the air near the earth is warmer than the air at greater heights, sometimes it is colder. Horizontal changes are taking place at the same time this vertical layering exists. All is a wondrously intricate and dynamic system, challenging the meteorologists (as well as acousticians) to make sense of it.

In the absence of thermal gradients, a sound ray may be propagated rectilinearly as shown in Fig. 12-3A. The sound ray concept is helpful in considering direction of propagation. Rays of sound are always perpendicular to sound wavefronts.

In Fig. 12-3B a thermal gradient exists between the cool air near the surface of the earth and the warmer air above. This affects the wavefronts of the sound. Sound travels faster in warm air than in cool air causing the tops of the wavefronts to go faster than the lower parts. The tilting of the wavefronts is such as to direct the sound rays downward. Under such conditions, sound from the source is bent down toward the surface of the earth and can be heard at relatively great distances.

The thermal gradient of Fig. 12-3C is reversed from that of Fig. 12-3B as the air near the surface of the earth is warmer than the air higher up. In this case the bottom parts of the wavefronts travel faster than the tops, resulting in an upward refraction of the sound rays. The same sound energy from the source *S* would now be dissipated in the upper reaches of the atmosphere, reducing the chances of it being heard at any great distance at the surface of the earth.

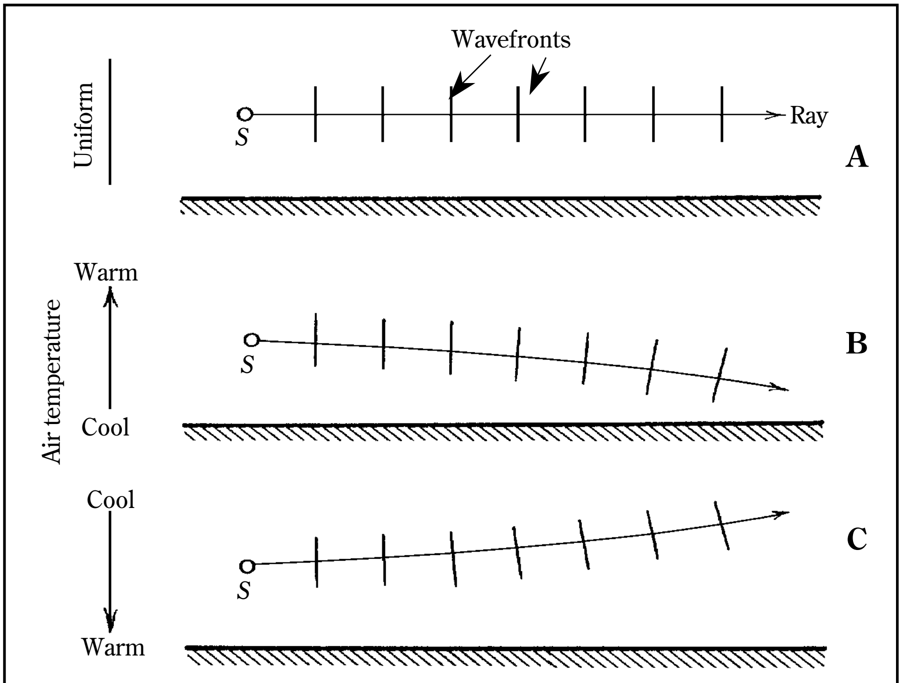

FIGURE 12-3

Refraction of sound paths resulting from temperature gradients in the atmosphere; (A) air temperature constant with height, (B) cool air near the surface of the earth and warmer air above, (C) warm air near the earth and cooler air above.

Figure 12-4A presents a distant view of the downward refraction situation of Fig. 12-3B. Sound traveling directly upward from the source S penetrates the temperature gradient at right angles and would not be refracted. It would speed up and slow down slightly as it penetrates the warmer and cooler layers, but would still travel in the vertical direction.

All rays of sound except the vertical would be refracted downward. The amount of this refraction varies materially: the rays closer to the vertical are refracted much less than those more or less parallel to the surface of the earth.

Figure 12-4B is a distant view of the upward refraction situation of Fig.12-3C. Shadow zones are to be expected in this case. Again, the vertical ray is the only one escaping refractive effects.

It is a common experience to hear sound better downwind than upwind. Air is the medium for the sound. If wind moves the air at a

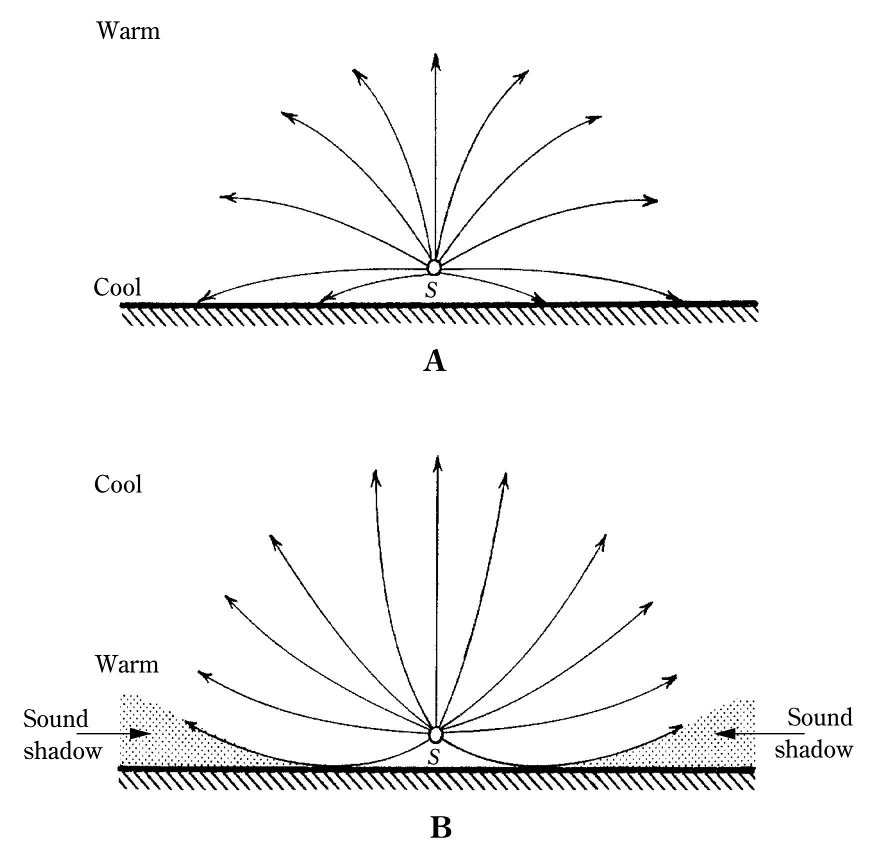

FIGURE 12-4

Comprehensive illustration of refraction of sound from source S; (A) cool air near the ground and warmer air above, (B) warm air near the ground and cooler air above. In (B) note that sound shadow areas result from the upward refraction.

certain speed, it is to be expected that the speed of sound will be affected. If sound travels 1,130 ft/sec and a 10 mi/hour (about 15 ft/sec) wind prevails, what will be the effect of the wind on the sound? Upwind the sound speed with respect to the earth would be increased about 1%, and downwind it would be decreased the same amount. This seems like a very small change but it is enough to affect refraction materially. Figure 12-5 illustrates the effect of wind on the downward refraction case of Fig. 12-4A. A downwind shadow is created and upwind sound is refracted downward.

Wind speed near the surface of the earth is usually less than that at greater heights. A wind gradient exists in such a case that has its effect on propagation of sound. This is not a true refraction but the effect is the same. Plane waves from a distant source traveling with the wind would bend the sound down toward the earth. Plane waves traveling against the wind will be bent upward.

It is possible, under unusual circumstances, that sound traveling upwind may actually be favored. For instance, upwind sound is kept above the surface of the ground, minimizing losses at the ground surface. After all, does not the sportsman approach his prey upwind? Doing so keeps footstep noises from being heard by the prey until the sportsman is quite close.

Refraction of Sound in the Ocean

In 1960 some oceanographers devised an ambitious plan to see how far underwater sound could be detected.[1,2] Charges of 600 lb were discharged at various depths in the ocean off Perth, Australia. Sounds from these discharges were detected near Bermuda. The great circle path the sound presumably followed is shown in Fig. 12-6. Even though sound in sea water travels 4.3 times faster than in air, it took 13,364 seconds (3.71 hr) for the sound to make the trip. This distance is over 12,000 miles, close to half the circumference of the earth. Interesting, but what has this to do with refraction? Everything!

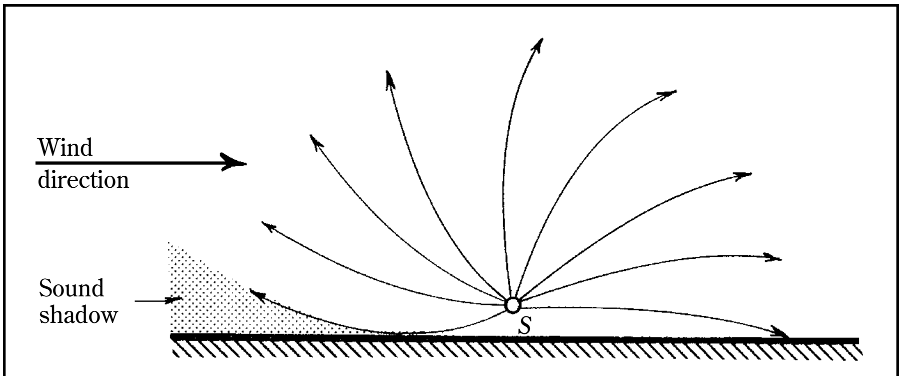

FIGURE 12-5

Wind gradients refract (not a true refraction) sound. A shadow sound is created upwind and good listening conditions downwind.

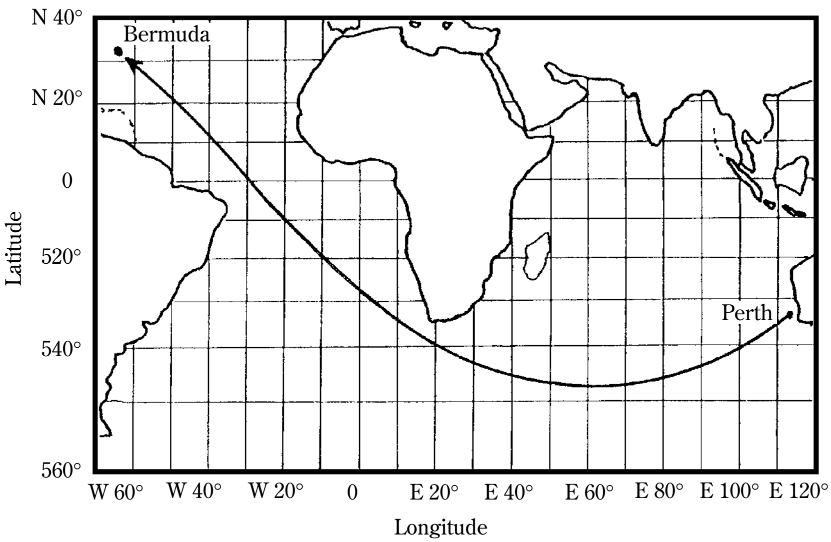

FIGURE 12-6

Refraction of sound in the ocean. A 600-lb charge was detonated near Perth, Australia, and the sound was recorded at Bermuda, over 12,000 miles away. The secret lies in the fact that the sound was confined to a sound channel by refraction that reduced losses. The sound took 3.71 hours to travel almost half way around the world. Such long-distance transmission of sound in the sea is being used to study long-range warming effects of the ocean. (After Heaney et al.[2])

An explanation is found in Fig. 12-7. The depth of the oceanic abyss is 5,000 or more fathoms (30,000 ft). At about 700 fathoms (4,200 ft) a very interesting effect takes place. The sound speed profile shown in Fig. 12-7A is very approximate to illustrate a principle. In the upper reaches of the ocean the speed of sound decreases with depth because temperature decreases. At greater depths the pressure effect prevails causing sound speed to increase with depth because of the increase in density. The "V" change-over from one effect to the other occurs near the 700 fathom (4,200 ft) depth.

A sound channel is created by this V-shaped sound-speed profile. A sound emitted in this channel tends to spread out in all directions. Any ray traveling upward will be refracted downward, any ray traveling downward will be refracted upward. Sound energy in this channel is propagated great distances with modest losses.

Refraction in the vertical plane is very prominent because of the vertical temperature/pressure gradient of Fig. 12-7A. There is relatively little

horizontal sound speed gradient and therefore very little horizontal refraction. Sound tends to be spread out in a thin sheet in this sound channel at about 700 fathom depth. Spherical divergence in three dimensions is changed to two-dimensional propagation at this special depth.

These long-distance sound channel experiments have suggested that such measurements can be used to check on the "warming of the planet" by detecting changes in the average temperature of the oceans. The speed of sound is a function of the temperature of the ocean. Accurate measures of time of transit over a given course yield information on the temperature of that ocean.[3]

Refraction of Sound in Enclosed Spaces

Refraction is an important effect on a world-sized scale, how about enclosed spaces? Consider a multi-use gymnasium that serves as an

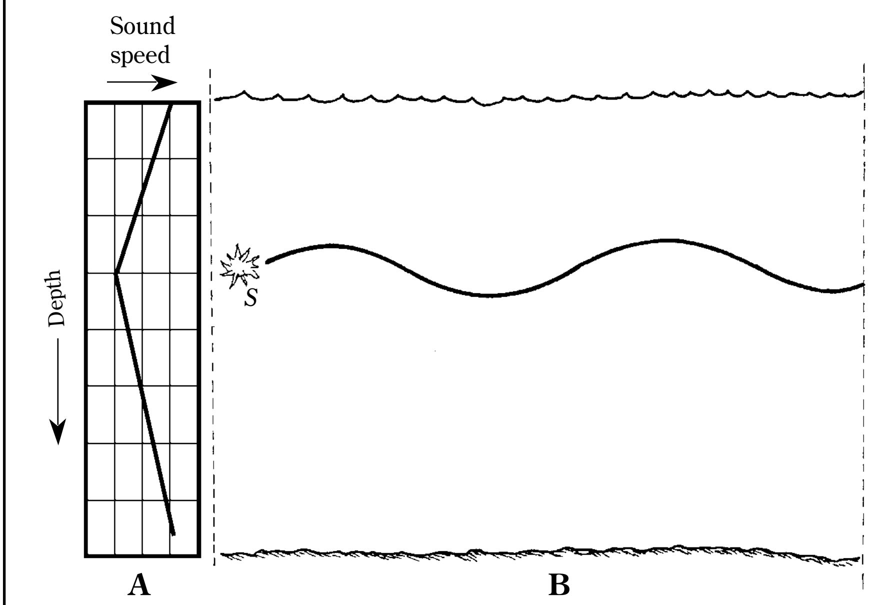

FIGURE 12-7

How the long range sound experiment of Fig. 12-6 was accomplished; (A) sound speed decreases with depth in the upper reaches of the ocean (temperature effect) and increases at greater depths (pressure effect) creating a sound channel at the inversion depth (about 700 fathoms). (B) A ray of sound is kept in this sound channel by refraction. Sound travels great distances in this channel because of the lower losses.

auditorium at times. With a normal heating and air-conditioning system, great efforts are made to avoid large horizontal or vertical temperature gradients. The goals of temperature uniformity and no troublesome drafts have reduced sound refraction effects to inconsequential levels.

Consider the same gymnasium used as an auditorium but with less sophisticated air conditioning. In this case a large ceiling-mounted heater near the rear acts as a space heater. Working against gravity, the unit produces copious hot air near the ceiling, relying on slow convection currents to move some of the heat down to the audience level.

This reservoir of hot air near the ceiling and cooler air below can have a minor effect on the transmission of sound from the sound system and on the acoustics of the space. The feedback point of the sound system might shift. The standing waves of the room might change slightly as longitudinal and transverse sound paths are increased in length because of their curvature due to refraction. Flutter echo paths are also shifted. With a sound radiating system mounted high at one end of the room, lengthwise sound paths would be curved downward. Such downward curvature might actually improve audience coverage, depending somewhat on the directivity of the radiating system.

Endnotes

[1]Shockley, R.C., J. Northrop, P.G. Hansen, and C. Hartdegen, *SOFAR Propagation Paths from Australia to Bermuda*, J. Acous. Soc. Am., 71, 51 (1982).

[2]Heaney, K.D., W.A. Kuperman, and B.E. McDonald, *Perth-Bermuda Sound Propagation (1960): Adiabatic Mode Interpretation*, J. Acous. Soc. Am., 90, 5 (Nov 1991) 2586-2594.

[3]Spiesberger, John, Kent Metzger, and John A. Ferguson, *Listening for Climatic Temperature Changes in the Northeast Pacific 1983-1989*, J. Acous. Soc. Am., 92, 1 (July 1992) 384-396.

CHAPTER 13

Diffusion of Sound

Diffusion problems are most troublesome in smaller rooms and at the lower audio frequencies. The problem with small spaces such as the average recording studio, control room, or music listening room is that modal spacings below 300 Hz guarantee a sound field far from diffuse (Chap. 15).

The Perfectly Diffuse Sound Field

Even though unattainable, it is instructive to consider the characteristics of a diffuse sound field. Randall and Ward[1] have given us a list of these:

- The frequency and spatial irregularities obtained from steady-state measurements must be negligible.
- Beats in the decay characteristic must be negligible.
- Decays must be perfectly exponential, i.e., they must be straight lines on a logarithmic scale.
- Reverberation time will be the same at all positions in the room.
- The character of the decay will be essentially the same for different frequencies.

- The character of the decay will be independent of the directional characteristics of the measuring microphone.

These six factors are observation oriented. A professional physicist specializing in acoustics might stress fundamental and basic factors in his definition of a diffuse sound field such as energy density, energy flow, superposition of an infinite number of plane progressive waves, and so on. The six characteristics suggested by Randall and Ward point us to practical ways of obtaining solid evidence for judging the diffuseness of the sound field of a given room.

Evaluating Diffusion in a Room

There is nothing quite as upsetting as viewing one's first attempt at measuring the "frequency response" of a room. To obtain the frequency response of an amplifier, a variable-frequency signal is put in the front end and the output observed to see how flat the response is. The same general approach can be applied to a room by injecting the variable frequency signal into "the front end" by means of a loudspeaker and noting the "output" picked up by a microphone located elsewhere in the room.

Steady-State Measurements

Figure 13-1 is a graphic-level recorder tracing of the steady-state response of a studio having a volume of 12,000 cubic feet. In this case, the loudspeaker was in one lower tricorner of the room, and the microphone was at the upper diagonal tricorner about one foot from each of the three surfaces. These positions were chosen because all room modes terminate in the corners and all modes should be represented in the trace. The fluctuations in this response cover a range of about 35 dB over the linear 30- to 250-Hz sweep. The nulls are very narrow, and the narrow peaks show evidence of being single modes because the mode bandwidth of this room is close to 4 Hz. The wider peaks are the combined effect of several adjacent modes. The rise from 30 to 50 Hz is due primarily to loudspeaker response and the 9-dB peak between 50 and 150 Hz (due to radiating into ¼ space) should not be charged against the room. The rest is primarily room effect.

The response of Fig. 13-1 is typical of even the best studios. Such variations in response are, of course, evidence of a sound field that is

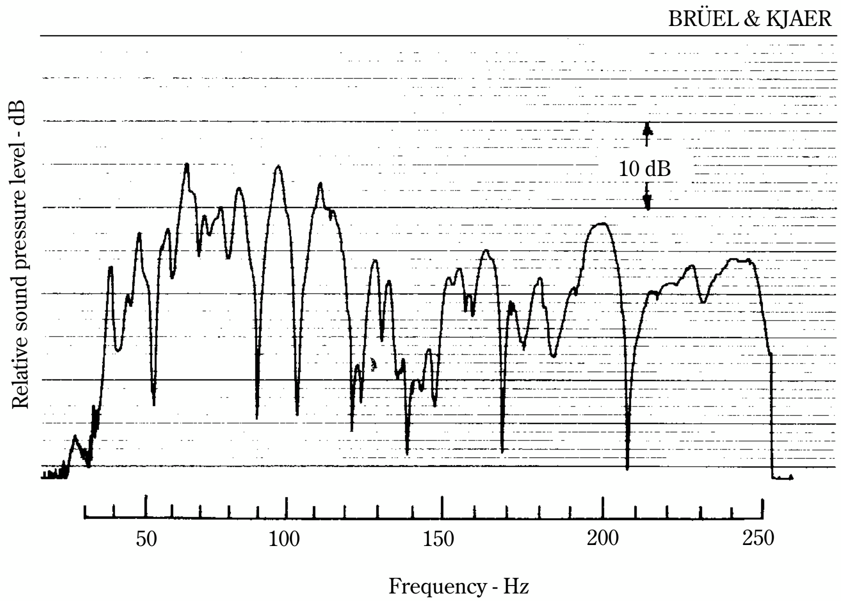

FIGURE 13-1

Slowly swept sine-wave sound-transmission response of a 12,000-cu ft. video studio. Fluctuations of this magnitude, which characterize the best of studios, are evidence of nondiffusive conditions.

not perfectly diffused. A steady-state response such as this taken in an anechoic room would still show variations, but of lower amplitude. A very live room, such as a reverberation chamber, would show even greater variations.

Figure 13-1 illustrates one way to obtain the steady-state response of a room. Another is to traverse the microphone while holding the loudspeaker frequency constant. Both methods reveal the same deviations from a truly homogeneous sound field. Thus, we see that Randall and Ward's criteria of negligible frequency and spatial irregularities are not met in the studio of Fig. 13-1 or, in fact, in any practical recording studio.

Decay Beats

By referring to Chap. 7, Fig. 7-10, we can compare the smoothness of the reverberation decay for the eight octaves from 63 Hz to 8 kHz. In

general, the smoothness of the decay increases as frequency is increased. The reason for this, as explained in Chap. 7, is that the number of modes within an octave span increases greatly with frequency, and the greater the mode density, the smoother their average effect. Beats in the decay are greatest at 63 Hz and 125 Hz. The decays of Fig. 7-10 indicate that the diffusion of sound in this particular studio is about as good as can be achieved by traditional means. It is the beat information on the low-frequency reverberation decay that makes possible a judgment on the degree of diffusion prevailing. Reverberation-time measuring devices that yield information only on the average slope and not the shape of the decay pass over information that most consultants consider important in evaluating the diffuseness of a space.

Exponential Decay

A truly exponential decay is a straight line on a level vs. time plot, and the slope of the line can be described either as a decay rate in decibels per second or as reverberation time in seconds. The decay of the 250-Hz octave band of noise pictured in Fig. 13-2 has two exponential slopes. The initial slope gives a reverberation time of 0.35 second and the final slope a reverberation time of 1.22 seconds. The slow decay that finally takes over once the level is low enough is probably a specific mode or group of modes encountering low absorption either by striking the absorbent at grazing angles or striking where there is little absorption. This is typical of one type of nonexponential decay, or stated more precisely, of a dual exponential decay.

Another type of nonexponential decay is illustrated in Fig. 13-3. The deviations from the straight line connecting the beginning and end of the decay are considerable. This is a decay of an octave band of noise centered on 250 Hz in a 400-seat chapel, poorly isolated from an adjoining room. Decays taken in the presence of acoustically coupled spaces are characteristically concave upward, such as in Fig. 13-3, and often the deviations from the straight line are even greater. When the decay traces are nonexponential, i.e., they depart from a straight line in a level vs. time plot, we must conclude that true diffuse conditions do not prevail.

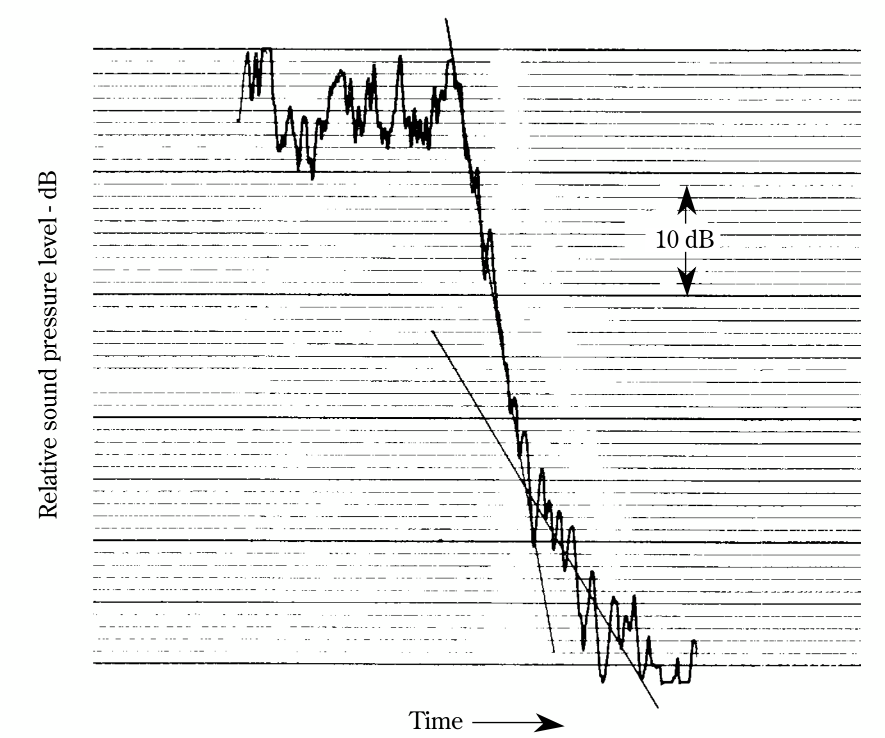

FIGURE 13-2

Typical double slope-decay, evidence of a lack of diffuse sound conditions. The slower decaying final slope is probably due to modes that encounter lower absorption.

Spatial Uniformity of Reverberation Time

When reverberation time for a given frequency is reported, it is usually the average of multiple observations at each of several positions in the room. This is the pragmatic way of admitting that reverberatory conditions differ from place to place in the room. Figure 13-4 shows the results of actual measurements in a small (22,000 cu ft) video studio. The multiple uses of the space required variable reverberation time, which was accomplished by hinged wall panels that can be closed, revealing absorbent sides, or opened, revealing reflecting sides. Multiple reverberation decays were recorded at the same three microphone positions for both "panels-reflective" and "panels-absorptive" conditions. The circles are the average values, and the light lines represent average reverberation time at each of the three positions. It is evident

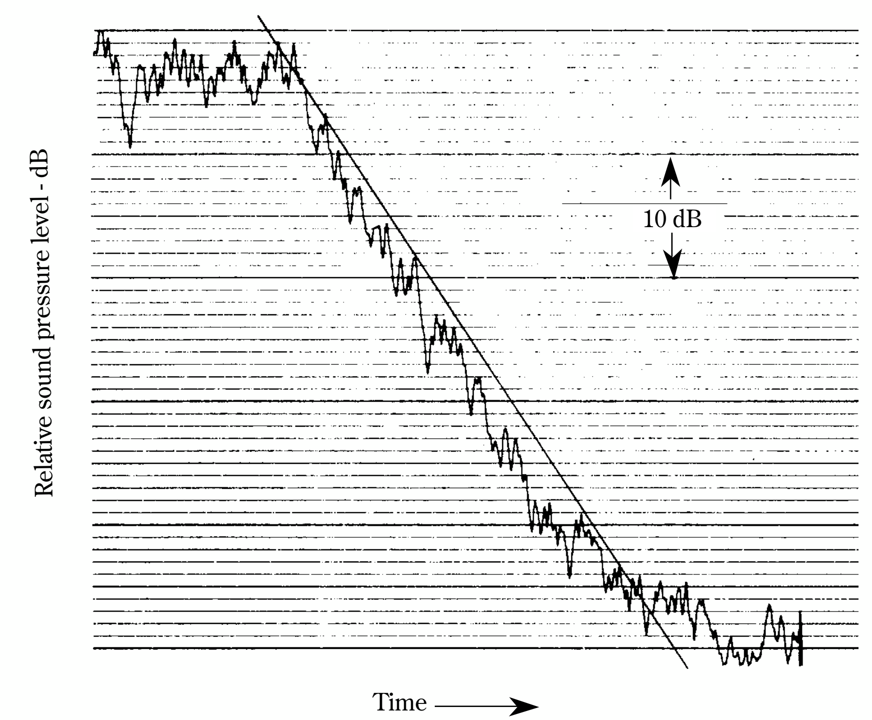

FIGURE 13-3

The nonexponential form of this decay, taken in a 400-seat chapel, is attributed to acoustically coupled spaces. The absence of a diffuse sound field is indicated.

that there is considerable variation, which means that the sound field of the room is not completely homogeneous during this transient decay period. Inhomogeneities of the sound field are one reason that reverberation times vary from point to point in the room, but there are other factors as well. Uncertainties in fitting a straight line to the decay also contribute to the spread of the data, but this effect should be relatively constant from one position to another. It seems reasonable to conclude that spatial variations in reverberation time are related, at least partially, to the degree of diffusion in the space.

Standard deviations of the reverberation times give us a measure of the spread of the data as measured at different positions in a room. When we calculate an average value, all evidence of the spread of the data going into the average is lost. The standard deviation is the statistician's way of keeping an eye on the data spread. The method of calcu-

lating the standard deviation is described in the manuals of most scientific calculators. Plus or minus one standard deviation from the mean value embraces 68% of the data points if the distribution is normal (Gaussian), and reverberation data should qualify reasonably well. In Table 13-1, for 500 Hz, panels reflective, the mean RT60 is 0.56 seconds with a standard deviation of 0.06 seconds. For a normal distribution, 68% of the data points would fall between 0.50 and 0.62 second. That 0.06 standard deviation is 11% of the 0.56 mean. The percentages listed in Table 13-1 give us a rough appraisal of the precision of the mean.

In order to view the columns of percentage in Table 13-1 graphically, they are plotted in Fig. 13-5. Variability of reverberation time values at the

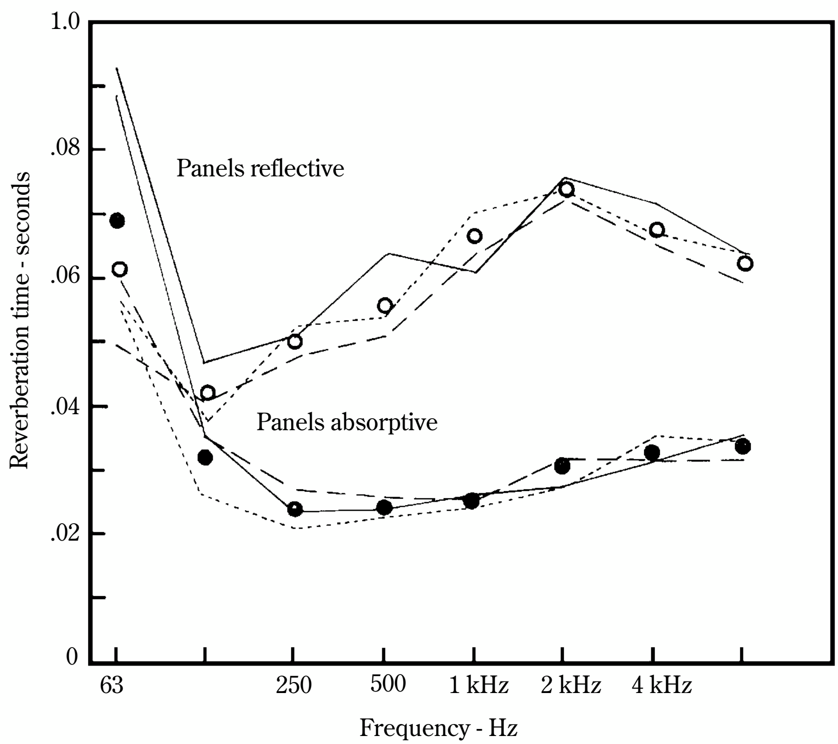

FIGURE 13-4

Reverberation time characteristics of a 22,000-cu ft studio with acoustics adjustable by hinged panels, absorbent on one side and reflective on the other. At each frequency, the variation of the average reverberation time at each of the three positions indicates non-diffuse conditions especially at low frequencies.

Table 13-1 Reverberation time of small video studio.

Octave band center frequency	Panels reflective RT60	Panels reflective Std. dev.	Panels reflective Std. dev. % of mean	Panels absorptive RT60	Panels absorptive Std. dev.	Panels absorptive Std. dev. % of mean
63	0.61	0.19	31.	0.69	0.18	26.
125	0.42	0.05	12.	0.32	0.06	19.
250	0.50	0.05	10.	0.24	0.02	8.
500	0.56	0.06	11.	0.24	0.01	4.
1 kHz	0.67	0.03	5.	0.26	0.01	4.
2 kHz	0.75	0.04	5.	0.31	0.02	7.
4 kHz	0.68	0.03	4.	0.33	0.02	6.
8 kHz	0.63	0.02	3.	0.34	0.02	6.

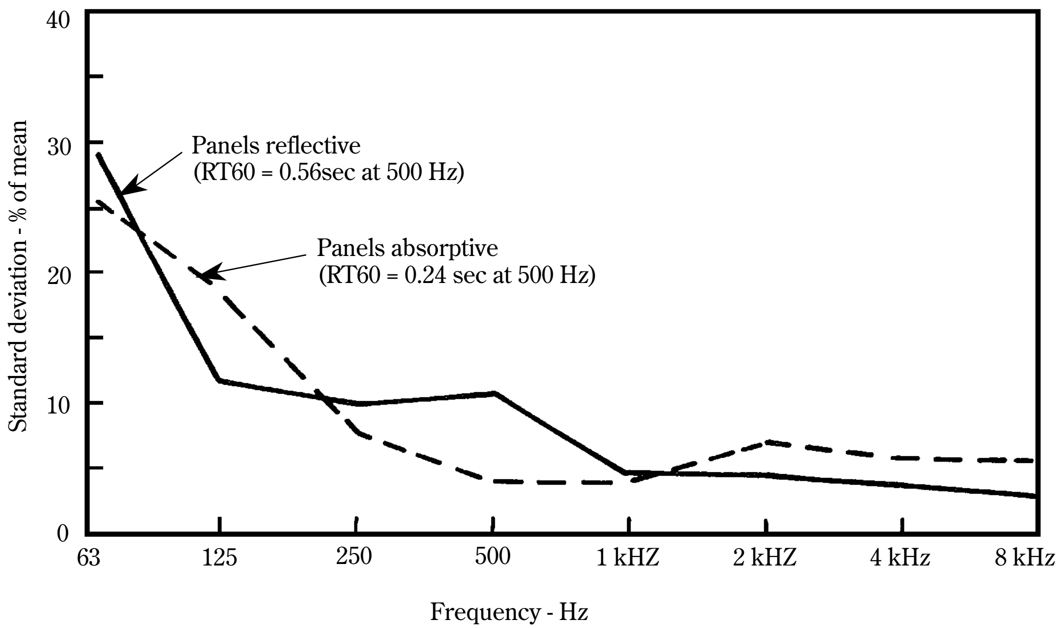

FIGURE 13-5

Closer examination of the reverberation time variations of the studio of Fig. 13-4. The standard deviation, expressed as a percentage of the mean value, shows lack of diffusion, especially below 250 Hz.

higher frequencies settles down to reasonably constant values in the neighborhood of 3% to 6%. Because we know that each octave at high frequencies contains an extremely large number of modes that results in smooth decays, we can conclude confidently that at the higher audible frequencies essentially diffuse conditions exist, and that the 3% to 6% variability is normal experimental measuring variation. At the low frequencies, however, the high percentages (high variabilities) are the result of greater mode spacing producing considerable variation in reverberation time from one position to another. We must also admit that these high percentages include the uncertainty in fitting a straight line to the wiggly decay characteristic of low frequencies. However, a glance at Fig. 13-4 shows that there are great differences in reverberation time between the three measuring positions. For this 22,000 cu ft studio for two different conditions of absorbance (panels open/closed), diffusion is poor at 63 Hz, somewhat better at 125 Hz, and reasonably good at 250 Hz and above.

Decay Shapes

If all decays have the same character at all frequencies and that character is smooth decay, complete diffusion prevails. In the real world, the decays of Fig. 7-10 with significant changes in character are more common, especially for the 63-Hz and 125-Hz decays.

Microphone Directivity

One method of appraising room diffusion is to rotate a highly directional microphone in various planes and record its output to the constant excitation of the room. This method has been applied with some success to large spaces, but the method is ill adapted to smaller recording studios, control rooms, and listening rooms, in which diffusion problems are greatest. In principle, however, in a totally homogeneous sound field, a highly directional microphone pointed in any direction should pick up a constant signal.

Room Shape

How can a room be built to achieve maximum diffusion of sound? This opens up a field in which there are strong opinions—some of them

supported by quite convincing experiments—and some just strong without such support.

There are many possible shapes of rooms. Aside from the general desirability of a flat floor in this gravity-stricken world, walls can be splayed, ceilings inclined, cylindrical or polygonal shapes employed. Some shapes can be eliminated because they focus sound, and focusing is the opposite of diffusing. For example, parabolic shapes yield beautifully sharp focal points and cylindrical concavities less sharp but nonetheless concentrated. Even polygonal concave walls of 4, 5, 6, or 8 sides approach a circle and result in concentrations of sound in some areas at the expense of others.

The popularity of rectangular rooms is due in part to economy of construction, but it has its acoustical advantages. The axial, tangential, and oblique modes can be calculated with reasonable effort and their distribution studied. For a first approximation, a good approach is to consider only the more dominant axial modes, which is a very simple calculation. Degeneracies (mode pile-ups) can be spotted and other room faults revealed.

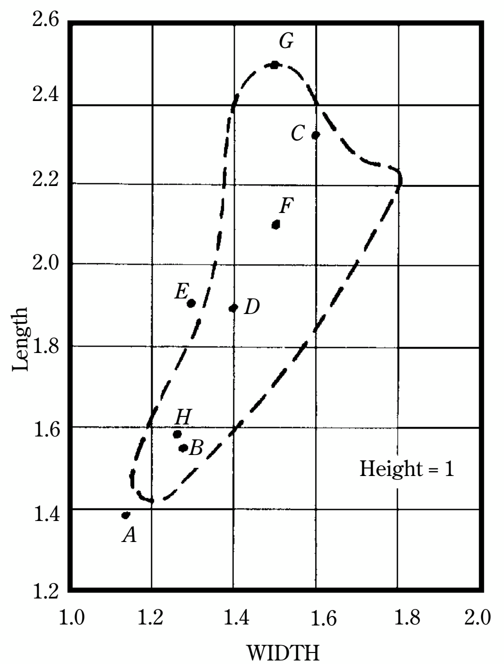

FIGURE 13-6

A chart of favorable room dimensional ratios to achieve uniform distribution of modal frequencies of a room. The broken line encloses the so-called "Bolt-Area."[2] The letters refer to Table 13-2.

The relative proportioning of length, width, and height of a sound sensitive room is most important. If plans are being made for constructing such a room, there are usually ideas on floor-space requirements, but where should one start in regard to room proportions? Cubical rooms are anathema. The literature is full of early quasi-scientific guesses, and later statistical analyses of room proportions that give good mode distribution. None of them come right out and say, "This is the absolute optimum." Bolt[2] gives a range of room proportions producing the smoothest room characteristics at low frequencies in small rectangular rooms (Fig. 13-6). Volkmann's 2 : 3 : 5 proportion,[3] was in favor 50 years ago. Boner suggested the 1 : 1.26 : 1.59 ratio as optimum.[4] Sepmeyer[5]

published a computer statistical study in 1965 that yields several favorable ratios. An even later paper by Louden[6] lists 125 dimension ratios arranged in descending order of room acoustical quality.

Table 13-2 lists the best proportions suggested by all of these papers. To compare these with the favorable area suggested by Bolt, they are plotted in Fig. 13-6. Most of the ratios fall on or very close to the *Bolt area*. This gives confidence that any ratio falling in the Bolt area will yield reasonable low-frequency room quality as far as distribution of axial modal frequencies is concerned.

One cannot tell by looking at a room's dimensional ratio whether it is desirable or not, and it is preferable to make the evaluation, rather than just take someone's word for it. Assuming a room height of 10 ft, and the other two dimensions, an axial mode analysis such as Fig. 13-7 can be made for each. This has been done and these modes are plotted in Fig. 13-8. Each is keyed into Table 13-2 for source identification. All of these are relatively small rooms and therefore suffer the same fate of having axial-mode spacings in frequency greater than desired. The more uniform the spacing, the better. Degeneracies, or mode coin-

Table 13-2. Rectangular room dimension ratios for favorable mode distribution.

Author		Height	Width	Length	In Bolt's range?
1. Sepmeyer[5]	A	1.00	1.14	1.39	No
	B	1.00	1.28	1.54	Yes
	C	1.00	1.60	2.33	Yes
2. Louden[6]	D	1.00	1.4	1.9	Yes
3 best ratios	E	1.00	1.3	1.9	No
	F	1.00	1.5	2.5	Yes
3. Volkmann[3]	G	1.00	1.5	2.5	Yes
2 : 3 : 5					
4. Boner[4]	H	1.00	1.26	1.59	Yes
$1: \sqrt[3]{2}: \sqrt[3]{4}$					

	Length $L = 19$ ft 5 in $L = 19.417$ ft $f_1 = 565/L$	Width $W = 14$ ft 2 in $W = 14.17$ ft $f_1 = 565/W$	Height $H = 8$ ft $H = 8.92$ ft $f_1 = 565/H$
f_1	29.1	39.9	63.3
f_2	58.1	79.7	126.7
f_3	87.3	119.6	190.0
f_4	116.4	159.5	253.4
f_5	145.5	199.4	316.7
f_6	174.6	239.2	
f_7	203.7	279.1	
f_8	232.8	319.0	
f_9	261.9		
f_{10}	291.0		
f_{11}	320.1		
f_{12}			
f_{13}			
f_{14}			
f_{15}			

Arranged in ascending order	Diff
29.1	
	10.8
39.9	
	18.3
58.2	
	5.1
63.3	
	16.4
79.7	
	7.6
87.3	
	29.1
116.4	
	3.2
119.6	
	7.1
126.7	
	18.8
145.5	
	14.0
159.5	
	15.1
174.6	
	15.4
190.0	
	9.4
199.4	
	4.3
203.7	
	29.1
232.8	
	6.4
239.2	
	14.2
253.4	
	8.5
261.9	
	17.2
279.1	
	11.9
291.0	
	25.7
316.7	

FIGURE 13-7

A convenient data form for studying the effects of room proportions on the distribution of axial modes.

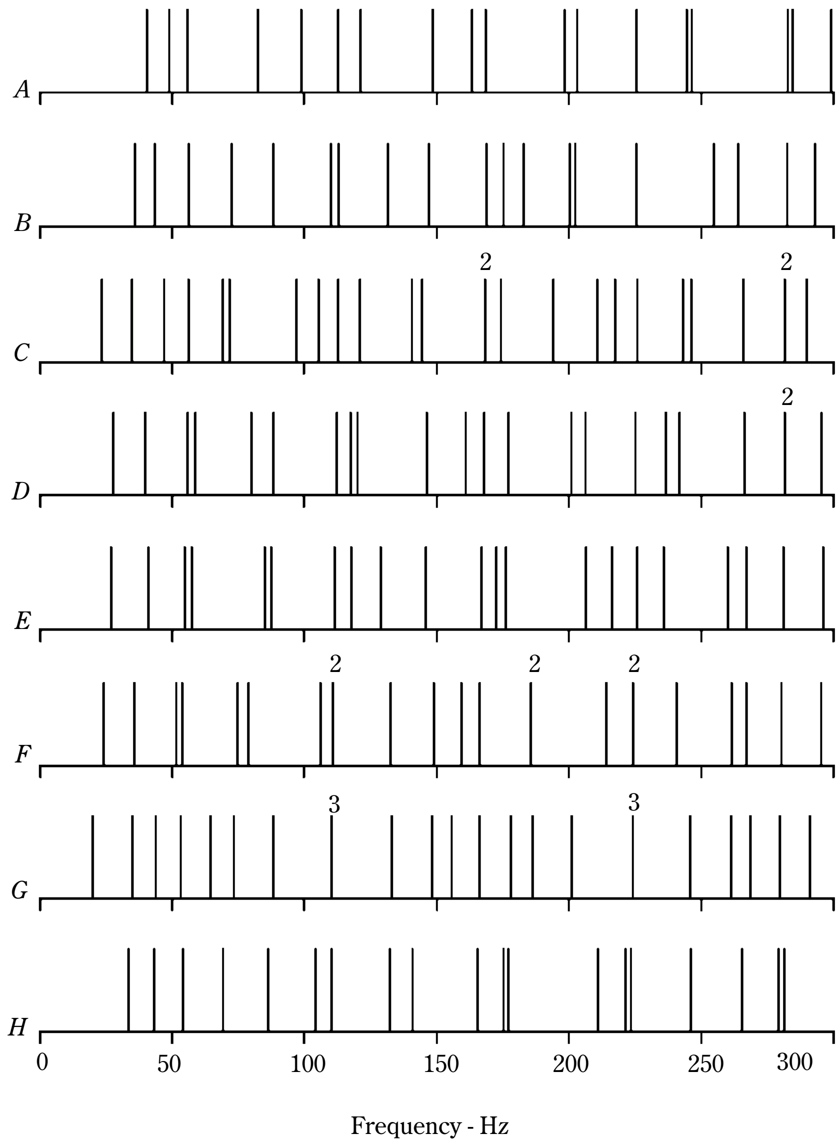

FIGURE 13-8

Plots of axial mode distribution for the eight "best" room proportions of Table 13-2. The small numbers indicate the number of modes coincident at those particular frequencies. A room height of 10 ft is assumed.

cidences, are a potential problem, and they are identified by the 2 or 3 above them to indicate the number of resonances piled up. Modes very close together, even though not actually coincident, can also present problems. With these rules to follow, which of the 8 "best" distributions of Fig. 13-8 are really the best and which the worst? First, we reject *G* with two triple coincidences greatly spaced from neighbors. Next, *F* is eliminated because of three double coincidences associated with some quite wide spacings. We can neglect the effect of the double coincidences near 280 Hz in *C* and *D* because colorations are rarely experienced above 200 Hz. Aside from the two rejected outright, there is little to choose between the remainder. All have flaws. All would probably serve quite well, alerted as we are to potential problems here and there. This simple approach of studying the axial-mode distribution has the advantage of paying attention to the dominant axial modes knowing that the weaker tangential and oblique modes can only help by filling in between the more widely spaced axial modes.

Figure 13-7 illustrates a data form that makes it easy to study the axial modes of a room. Analyzing the results requires some experience and a few rules of thumb are suggested. A primary goal is to avoid coincidences (pile-ups) of axial modes. For example, if a cubical space were analyzed, all three columns would be identical; the three fundamentals and all harmonics would coincide. This produces a triple coincidence at each modal frequency and great gaps between. Unquestionably, sound in such a cubical space would be highly colored and acoustically very poor. The room of Fig. 13-7 has 22 axial modes between 29.1 and 316.7 Hz. If evenly spaced, the spacing would be about 13 Hz, but spacings vary from 3.2 to 29.1 Hz. However, there are no coincidences—the closest pair are 3.2 Hz apart. If a new room is to be constructed, you have the freedom on paper to move a wall this way or that or to raise or lower the ceiling a bit to improve distribution. The particular room proportions of Fig. 13-7 represent the end product of many hours of cut and try. While this cannot be represented as the best proportioning possible, this room, properly treated, will yield good, uncolored sound. The proper starting point is proper room proportions.

In adapting an existing space, you lack the freedom to shift walls as on paper. A study of the axial modes as per Fig. 13-7, however, can still be very helpful. For example, if such a study reveals problems and space permits, a new wall might improve the modal situation

markedly. By splaying this wall, other advantages discussed later may accrue. If the study points to a coincidence at 158 Hz, well separated from neighbors, one is alerted to potential future problems with an understanding of the cause. There is always the possibility of introducing a Helmholtz resonator tuned to the offending coincidence to control its effect (see pp. 226–229). All these things are related to sound diffusion.

Splaying Room Surfaces

Splaying one or two walls of a sound-sensitive room does not eliminate modal problems, although it might shift them slightly and produce somewhat better diffusion.[7] In new construction, splayed walls cost no more, but may be quite expensive in adapting an existing space. Wall splaying is one way to improve general room diffusion, although its effect is nominal. Flutter echoes definitely can be controlled by canting one of two opposing walls. The amount of splaying is usually between 1 foot in 20 feet and 1 foot in 10 feet.

Nonrectangular Rooms

The acoustical benefit to be derived from the use of nonrectangular shapes in audio rooms is rather controversial. Gilford[10] states, "...slanting the walls to avoid parallel surfaces.... does not remove colorations; it only makes them more difficult to predict." Massive trapezoidal shaped spaces, commonly used as the outer shell of recording studio control rooms, guarantee asymmetrical low-frequency sound fields even though it is generally conceded that symmetry with the control position is desirable.

Computer studies based on the finite element approach have revealed in minute detail what happens to a low-frequency sound field in nonrectangular rooms. The results of a study using this method conducted by van Nieuwland and Weber at Philips Research Laboratories, The Netherlands, are given in Figs. 13-10 through 13-13.[8] Highly contorted sound fields are shown, as expected, for the nonrectangular case, for modes 1,0, 1,3, 0,4, and 3,0. A shift in frequency of the standing wave from that of the rectangular room of the same area is indicated: –8.6%, –5.4%, –2.8%, and +1% in the four cases illustrated. This would tend to support the common statement that splay-

FIGURE 13-9

The use of distributed sound-absorbing modules is an economical way to achieve maximum absorption as well as to enhance sound diffusion in the room. *(World Vision International)*

ing of walls helps slightly in breaking up degeneracies, but shifts of 5% or more are needed to avoid the effects of degeneracies. The proportions of a rectangular room can be selected to eliminate, or at least greatly reduce, degeneracies, while in the case of the nonrectangular room, such a prior examination of degeneracies is completely impractical. Making the sound field asymmetrical by splaying walls only introduces unpredictability in listening room and studio situations.

If the decision is made to splay walls in an audio room, say 5%, a reasonable approximation would be to analyze the equivalent rectangular room having the same volume.

Geometrical Irregularities

Many studies have been made on what type of wall protuberances provide the best diffusing effect. Somerville and Ward[9] reported years ago that geometrical diffusing elements reduced fluctuations in a swept-sine steady-state transmission test. The depth of such geometrical diffusors must be at least ⅐ of a wavelength before their effect is felt. They studied cylindrical, triangular, and rectangular elements and found that the straight sides of the rectangular-shaped diffusor provided the greatest effect for both steady-state and transient phenomena. BBC experience indicates superior subjective acoustical properties in studios and concert halls in which rectangular ornamentation in the form of coffering is used extensively.

Absorbent in Patches

Applying all the absorbent in a room on one or two surfaces does not result in a diffuse condition, nor is the absorbent used most effectively. Let us consider the results of an experiment that shows the effect of

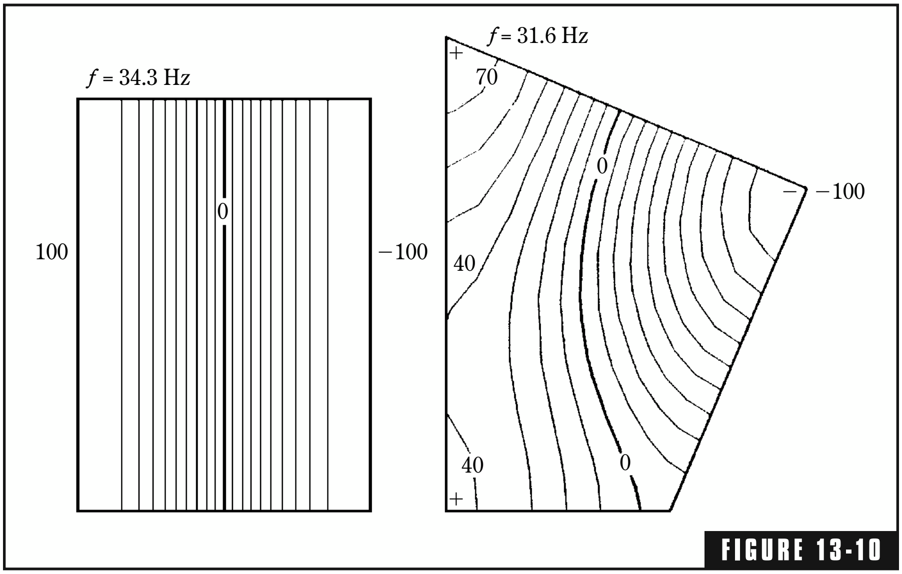

FIGURE 13-10

Comparison of the modal pattern for a 5 x 7 meter two-dimensional room and a nonrectangular room of the same area. This sound field of the 1,0 mode is distorted in the nonrectangular room and the frequency of the standing wave is shifted slightly.[8]

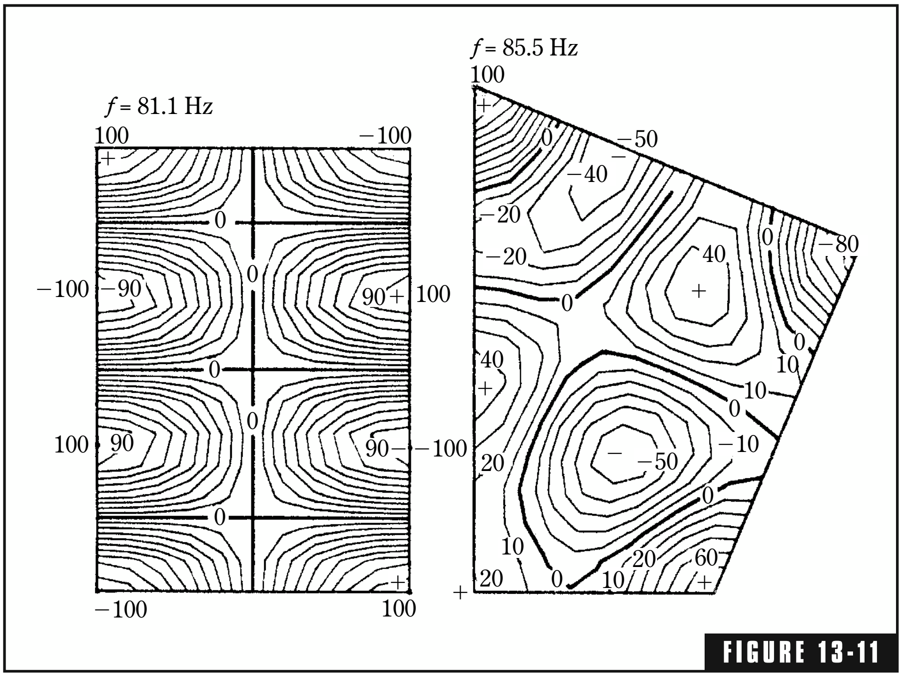

FIGURE 13-11

The 1,3 mode for the 5 x 7 meter room of Fig. 13-10 compared to a nonrectangular room of the same area. The sound field is distorted and the frequency is shifted.[8]

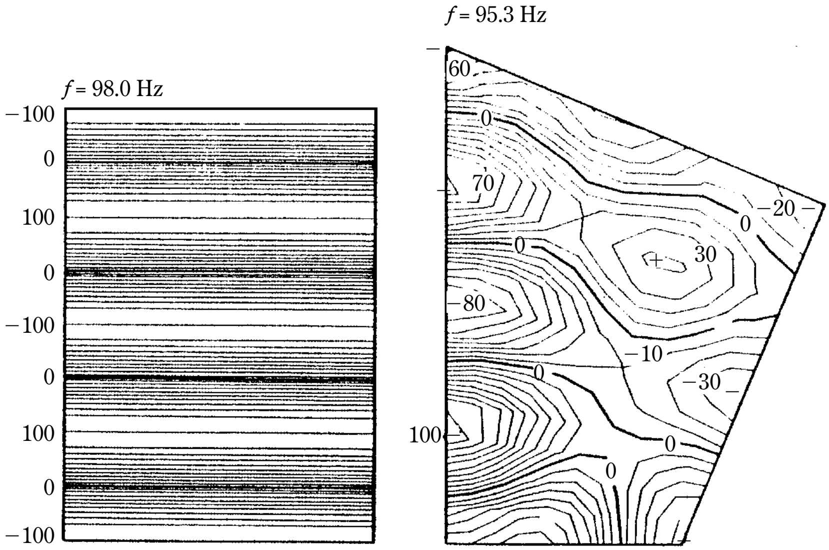

FIGURE 13-12

The 0,4 mode of the 5 x 7 meter two-dimensional room of Figs. 13-10 and 13-11. The high distortion in the rectangular room is accompanied by a shift in standing wave frequency.[8]

distributing the absorbent.[1] The experimental room is approximately a 10-ft cube and it was tiled (not an ideal recording or listening room, but acceptable for this experiment). For test 1, reverberation time for the bare room was measured and found to be 1.65 seconds at 2 kHz. For test 2, a common commercial absorber was applied to 65% of one wall (65 sq ft), and the reverberation time at the same frequency was found to be about 1.02 seconds. For test 3, the same area of absorber was divided into four sections, one piece mounted on each of four of the room's six surfaces. This brought the reverberation time down to about 0.55 seconds.

The startling revelation here is that the area of the absorber was identical between tests 2 and 3; the only difference was that in test 3 it was in four pieces, one on each of 3 walls and one piece on the floor. By the simple expedient of dividing the absorbent and distributing it, the reverberation time was cut almost in half. Inserting the values of reverberation time of 1.02 and 0.55 seconds and the volume and area of the room into the Sabine equation, we find that the average absorp-

tion coefficient of the room increased from 0.08 to 0.15 and the number of absorption units from 48 to 89 sabins. Where did all this extra absorption come from? Laboratory testing personnel measuring absorption coefficients in reverberation chambers have agonized over the problem for years. Their conclusion is that there is an edge effect related to diffraction of sound that makes a given sample appear to be much larger acoustically. Stated another way, the sound-absorbing efficiency of 65 sq ft of absorbing material is only about half that of four 16-sq ft pieces distributed about the room, and the edges of the four pieces total about twice that of the single 65-sq ft piece. So, one advantage of distributing the absorbent in a room is that its sound-absorbing efficiency is greatly increased, at least at certain frequencies. But be warned: The above statements are true for 2 kHz, but at 700 Hz and 8 kHz, the difference between one large piece and four distributed pieces is small.

Another significant result of distributing the absorbent is that it contributes to diffusion of sound. Patches of absorbent with reflective

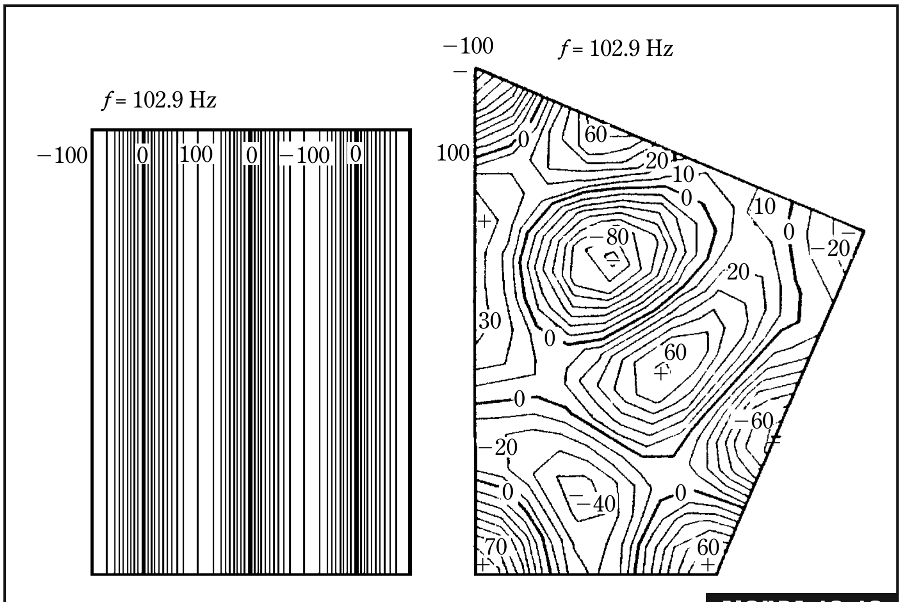

FIGURE 13-13

The 3,0 mode of the 5 x 7 meter two-dimensional room of Figs. 13-10 and 13-11, and 13-12 and resulting distortion of the modal pattern when changed to a nonrectangular room of the same area.[8]

walls showing between the patches have the effect of altering wavefronts, which improves diffusion. Sound-absorbing modules in a recording studio such as in Fig. 13-9 distribute the absorbing material and simultaneously contribute to the diffusion of sound.

Concave Surfaces

A concave surface such as that in Fig. 13-14A tends to focus sound energy and consequently should be avoided because focusing is just the opposite of the diffusion we are seeking. The radius of curvature determines the focal distance; the flatter the concave surface, the greater the distance at which sound is concentrated. Such surfaces often cause problems in microphone placement. Concave surfaces might produce some awe-inspiring effects in a whispering gallery where you can hear a pin drop 100 ft away, but they are to be avoided in listening rooms and small studios.

Convex Surfaces: The Poly

One of the most effective diffusing elements, and one relatively easy to construct, is the polycylindrical diffusor (poly), which presents a convex section of a cylinder. Three things can happen to sound falling on

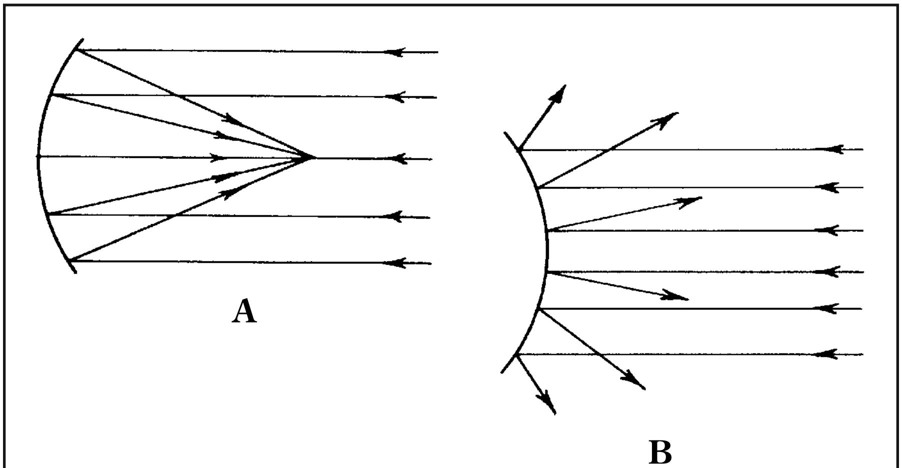

FIGURE 13-14

Concave surfaces (A) tend to focus sound, convex surfaces (B) tend to diffuse it. Concave surfaces should be avoided if the goal is to achieve well-diffused sound.

such a cylindrical surface made of plywood or hardboard: The sound can be reflected and thereby dispersed as in Fig. 13-14B; the sound can be absorbed; or the sound can be reradiated. Such cylindrical elements lend themselves to serving as absorbers in the low frequency range where absorption and diffusion are so badly needed in small rooms. The reradiated portion, because of the diaphragm action, is radiated almost equally throughout an angle of roughly 120° as shown in Fig. 13-15A. A similar flat element reradiates sound in a much narrower angle, about 20°. Therefore, favorable reflection, absorption, and reradiation characteristics favor the use of the cylindrical surface. Some very practical polys and their absorption characteristics are presented in Chap. 9. The dimensions of such diffusors are not critical, although to be effective their size must be comparable to the wavelength of the sound being considered. The wavelength of sound at 1,000 Hz is a bit over 1 ft, at 100 Hz about 11 ft. A poly element 3 or 4 ft across would be effective at 1000 Hz, much less so at 100 Hz. In general, poly base or chord length of 2 to 6 ft with depths of 6 to 18 inches meet most needs.

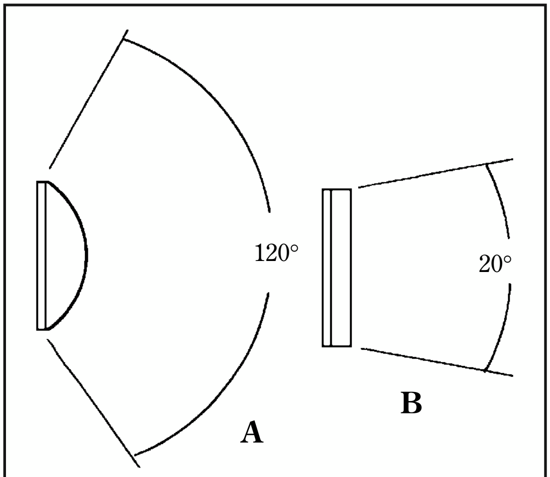

FIGURE 13-15

(A) A polycylindrical diffusor reradiates sound energy not absorbed through an angle of about 120°. (B) A similar flat element reradiates sound in a much smaller angle.

It is important that diffusing elements be characterized by randomness. A wall full of polys, all of 2-ft chord and of the same depth, might be beautiful to behold, like some giant washboard, but not very effective as diffusors. The regularity of the structure would cause it to act as a diffraction grating, affecting one particular frequency in a much different way than other frequencies, which is opposite to what the ideal diffusor should do.

Axes of symmetry of the polys on different room surfaces should be mutually perpendicular.

Plane Surfaces

Geometrical sound diffusing elements made up of two flat surfaces to give a triangular cross section or of three or four flat surfaces to give a

polygonal cross section may also be used. In general, their diffusing qualities are inferior to the cylindrical section.

Endnotes

[1]Randall, K.E. and F.L. Ward, *Diffusion of Sound in Small Rooms*, Proc. Inst. Elect. Engs., Vol 107B (Sept. 1960), p. 439-450.

[2]Bolt, R.H., *Note on Normal Frequency Statistics for Rectangular Rooms*, J. Acous. Soc. Am., 18, 1 (July 1946) p. 130-133.

[3]Volkmann, J.E., *Polycylindrical Diffusers in Room Acoustical Design*, J. Acous. Soc. Am., 13 (1942), p. 234-243.

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[5]Sepmeyer, L.W., *Computed Frequency and Angular Distribution of the Normal Modes of Vibration in Rectangular Rooms*, J. Acous Soc. Am., 37, 3 (March 1965), p. 413-423.

[6]Louden, M.M., *Dimension-Ratios of Rectangular Rooms with Good Distribution of Eigentones*, Acustica, 24 (1971), p. 101-103.

[7]Nimura, Tadamoto and Kimio Shibayama, *Effect of Splayed Walls of a Room on Steady-State Sound Transmission Characteristics*, J. Acous. Soc. Am., 29, 1 (January 1957), p. 85-93.

[8]van Nieuwland, J.M. and C. Weber, *Eigenmodes in Non-Rectangular Reverberation Rooms*, Noise Control Eng., 13, 3 (Nov/Dec 1979), 112-121.

[9]Somerville, T. and F.L. Ward, *Investigation of Sound Diffusion in Rooms by Means of a Model*, Acustica, 1, 1 (1951), p. 40-48.

[10]Gilford, Christopher, *Acoustics for Radio and Television Studios*, (1972), London, Peter Peregrinus, Ltd.